Operating manual
AM100

Bedrock AM100 operating manual

Contact information and support

The Bedrock AM100 is designed and manufactured by Bedrock Audio B.V. in the Netherlands.

The manufacturer can be contacted at the following address:

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Updates of the system's firmware and documentation are released online :

> www.bedrock-audio.com/support

NOTE: All screenshots and product pictures shown in this manual correspond to the AM100 running firmware version 3.2.3, hardware revision A. If you are using a different device model or a different firmware version, the screenshots may not fully correspond to what is being displayed on your device's screen.

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Contents

1. Introduction

Congratulations on purchasing your Bedrock measuring instrument! The Bedrock AM100 is our flagship instrument, featuring the latest technology and complying with the most recent standards.

Using the AM100 is quite easy, thanks to its touch-screen operation and intuitive menu structure. Most users will get the hang of doing measurements with the AM100 without spending too much time with this manual. Nonetheless, we strongly recommend that you do read chapters 1-5 of this manual before using your AM100 for the first time, in order to get acquainted with the various features of the device, as well as the procedures for charging and maintenance. With a little care, you can expect years and years of reliable and trouble-free operation from the Bedrock AM100.

Class 1 / type 1 microphone

Rubber acoustic diffuser ring

Power button

Touch screen

2nd input channel

Charge status LED

USB charge & data port

Headset/ generator jack
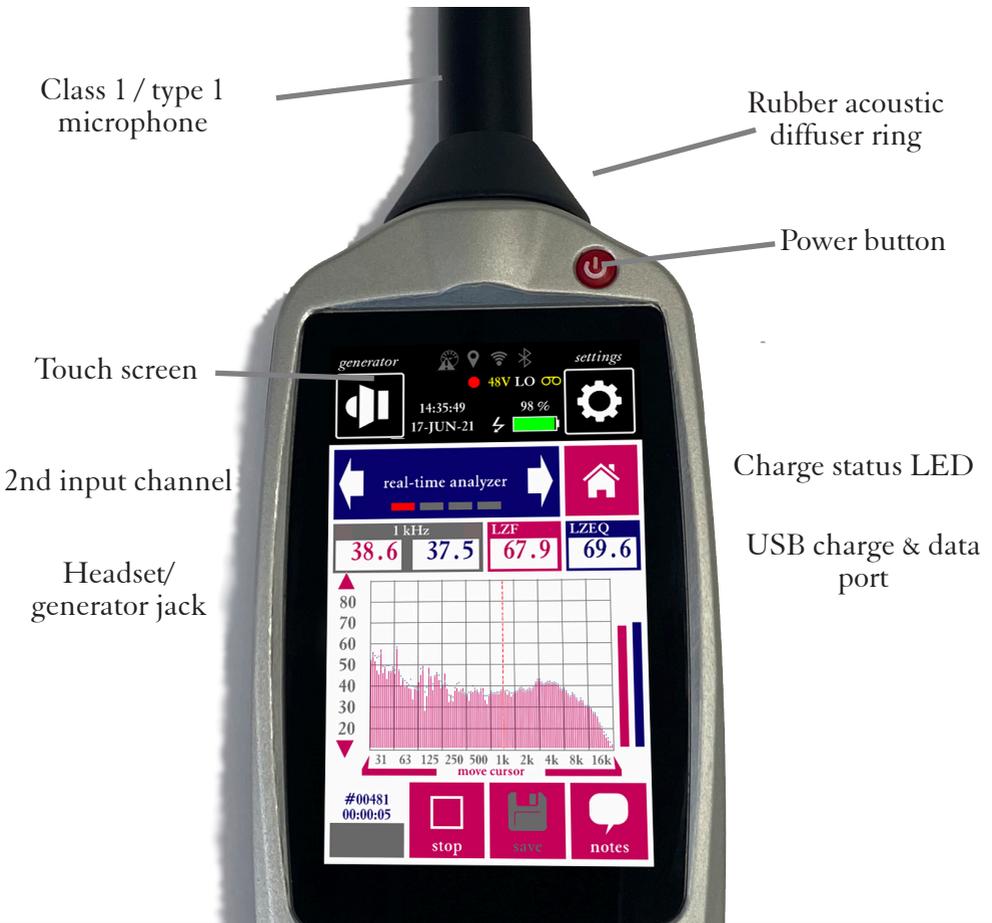
2. New features compared other Bedrock instruments (SMxx)

Users who have previously worked with the Bedrock SM30, SM50 or SM90 will experience a smooth and seamless transition to the Bedrock AM100. The user interface will appear very familiar and most menus and controls are found in approximately the same places as on the SMxx. All the modules, methods and features previously offered on the Bedrock SM90 are also found on the AM100. Nevertheless, the AM100 has been significantly improved and expanded in various ways:

New features
- Second simultaneous input channel, e.g. for frequency transfer function measurements.
- Headset connectivity, to listen in on the measurement microphone and to record voice annotations with your measurements.
- Built-in signal generator for STI signals, noise, sine waves and sweeps.
- Elaborate self-test capabilities
- Transfer data and recorded audio via WiFi.
- Remote control / remote operation via Bluetooth.
- (optional) built-in GPS to store locations for each measurement .

Improved specifications
- Improved electro-acoustic specs, including a greater dynamic range and an exceptionally flat frequency response.
- High-resolution capacitive touch screen.
- Extremely sturdy aerospace-grade aluminium enclosure.
- Improved user interface; control the most important settings through drop-down menus without having to switch modules.
- Improved battery life.
- Better overall battery management (e.g. more accurate battery gauge, ability to continue charging while measuring).
- USB-C connector capable of USB PD3.1 fast charge modes.
- Faster data transfer and firmware updates.

The Bedrock AM100 shares its output file formats with the rest of the Bedrock series, meaning that our Excel tools and any customized spreadsheets or existing third-party software will work with the AM100 as well.

3. How to use this manual

Depending on how you plan to use your Bedrock AM100, not all information in this manual may be equally interesting or relevant. If you are new to the Bedrock family of acoustic measuring instruments, we strongly recommend that you read the following chapters:

- Chapter 4: General use and safety precautions
- Chapter 5: Getting started
- Chapters on the measurement modules that you plan to use first
- Chapter 16: Power and charging
- Chapter 17: Cleaning, maintenance and calibration

We recommend reading the chapters on the various measurement modules as you plan to use them. As some of the modules implement complex measurement procedures, using those modules may require you to spend some time with this manual, and possibly also third-party source of information (such as online tutorials and course manuals).

The following sections should be read if you are experiencing issues with your device or if you need to upgrade your firmware:

- Before getting support from your Bedrock audio vendor, please consult the Troubleshooting section (chapter 22)
- Consult the section on Firmware updates (chapter 19) before trying to install new firmware on your device. Failure to follow firmware update instructions precisely may render your device permanently inoperable and voids your warranty.

Towards the end of this manual, you will find detailed information in tables and appendices that may not be needed for everyday use of the device. In particular, these appendices contain procedures for periodic testing, as needed by 3rd party labs for calibration and (re-) certification of your instrument.

New in firmware version 3.2

Firmware version 3.2 adds two new modules in the building acoustics menu, both intended for measuring the transmission of impact noise. It also adds a module in the building acoustics menu intended for Service Equipment Sounds measurements (SES) and two specific logging modules in the logging menu: Environment and Safety (E&S) and DIN-15905-5.

4. General use and safety precautions

Your Bedrock AM100 is a sophisticated electronic measuring instrument that should be used, maintained and stored with care:

- If you use a tripod to mount the device during measurements, we recommend that you use a sturdy model with a sufficiently wide base.
- Do not expose the device itself, the measuring microphone or charger to rain, moisture or liquids of any kind. Take special care to prevent liquids to permeate into the device through the connectors or along the display bezel.
- Do not operate or charge at ambient temperatures above 35°C/95F.
- Do not use close to flames or open fire.
- Do not use in environments where flammable or explosive materials may also be used.
- Do not attempt to charge at ambient temperatures above 35°C/95F or below 8°C/46F. The AM100 will block charging at these temperatures to prevent damage to the NiMH battery cells.
- Use the charger only with the international adapter suitable for your region.
- Make sure that the device is switched off before transportation and storage.
- During transportation and storage, keep the device in a suitable casing or container – preferably its original case. The device needs to be protected from shocks and vibration (due to transportation or falling), excessive temperatures, liquids and moisture, and any other external conditions that could do damage to the device. Care should be taken that other objects cannot come into direct contact with the display during transportation.

4.1. Safety precautions when charging

The Bedrock AM100 is a battery-powered rechargeable device, with an internal low self-discharge NiMH battery pack. This battery pack is designed for many years of operation, and therefore cannot be swapped out by the user. Recharging takes place through the mini USB connector found on the left side of the device.

The device is charged through the USB connector using high currents of up to 2100 mA. While being charged, it will heat up. DO NOT CHARGE THE DEVICE WHILE IT IS INSIDE A CLOSED CONTAINER (BOX, BRIEFCASE, ETC.). Please make sure that the device can shed excess heat while charging.

The supplied charger includes adapters for use worldwide, and operates on a wide AC voltage range of 100 – 240V. Do not use the charger if the casing appears cracked or broken, or if it has been exposed to fluids or moisture. Contact with components inside the charger while this is plugged in may result in serious injury or death.

4.2. Safety precautions related to connectors

The Bedrock AM100 features a USB connector, an XLR connector (to which the microphone is attached), and two TRRS connectors (signal generator/headset and 2nd input channel). If you connect external devices to these connectors, please ensure that these connections are protected against overvoltage and power surges. The voltage supplied to the USB port may not exceed 5.5V., except when USB PD2 or PD3 compliant fast chargers are used. Always use the original cables supplied with your AM100.

The XLR connector supplies 48V phantom power to the microphone. This phantom voltage can be switched on and off through the hardware settings menu. IMPORTANT: connecting equipment not designed for 48V phantom power to the XLR connector (with phantom power switched on) may permanently damage the connected device and/or the AM100. The manufacturer explicitly denounces responsibility for any damages to third-party hardware resulting from exposure to the Bedrock AM100's phantom power.

The "AC Volt meter" function of the device is intended for measuring audio output levels of amplifiers and other audio equipment. NEVER connect the XLR input pins to AC voltages over 40V; do NOT try to measure the AC mains with your AM100.

5. Getting started

You will most likely receive your Bedrock AM100 with sufficiently charged batteries to start the device and get acquainted with its features. We do recommend that you fully charge the device before you begin your first serious measurement session. **Leave the device on the charger until it finishes its charge cycle.** This may take up to 6 hours; many users opt for overnight charging. It is normal for the device to noticeably warm up during charging.

All rechargeable batteries slowly lose charge during storage, the batteries in the Bedrock AM100 are no exception. When you take out the device after it has been on the shelf for more than a few weeks, always fully charge the batteries first. The battery charge indicator may not be accurate after longer periods of storage.

5.1. Preparing and booting the AM100

Connect the microphone using the XLR connector at the top of the device. Optionally, for sensititive measurements, mount the rubber diffuser ring to reduce reflections off of the top of the AM100. Next, power on the device by pressing the red power button once.

Note: the power button is also used to power the device off. If you press the button while the device is in operation, you will see a dialog window asking confirmation. If the power button is kept pressed for 10 seconds in the "power on" state, the device will be forced into the "off" condition. This can be used to reset the device if it becomes unresponsive.

5.2. The AM100 main menu

The Bedrock AM100 will boot into the main menu. You can launch its various measurement modules from this menu.

The main menu is where all the available modules are shown, and where you can choose which module to launch. New modules are regularly added through firmware updates that are made available through the Bedrock website. If your device is running a different firmware version than the version on which this manual is based (version 3.0), you may see a different selection of modules appear in the menu than the ones found in this manual.
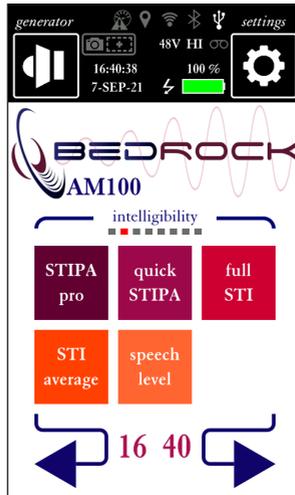
The above screenshot of the main menu shows its second page (speech intelligibility). Use the arrows to flip through the pages of the main menu. You will be able to choose from the following options:

Set menu favorites: press to select favorite modules to appear on this page

Intelligibility
STIPA pro: gives you access to all STI measurement details
Quick STIPA: quickly do STIPA measurements
Full STI: measure STI based on the complete 14x7 MTF-matrix
STI average: Take the average across multiple STIPA measurements
Speech level: measure speech level corrected for pauses (IEC-60268-16)

Level and spectrum
SPL meter: Sound Pressure Level meter module
LAEQ logging: long-term monitoring and logging of sound levels
RTA: Real-time analyzer
FFT: Fast Fourier Transform spectral analyzer
L%: calculate statistical levels such as L10 and L95 (percentiles)
RTA logging: Long-term monitoring of spectra

Room acoustics
RT60: Reverberation Time meter
Noise curves: Measure curve-based spectral single-number ratings (NR, NC, etc).

Building acoustics
Sound Insulation: In situ measurement of sound insulation between adjacent spaces
SRI / STC: Single-number insulation metrics

Electro-acoustics
Oscilloscope: Inspect audio waveforms in real time
Polarity: Check loudspeaker polarity
el.in RTA, el.in FFT: RTA and FFT for electric input signals via XLR

Electric analysis
AC Volts: Measure output levels of audio equipment in dBU, dBV and Vrms
Hz: Estimate the fundamental frequency of a periodic signal
THD+N: Total Harmonic Distortion and Noise

Settings and tools
Calibration: calibrate the AM100 and its microphone
Settings: change the system's hardware configuration
Network: change WiFi settings

The menu consists of multiple screens, through which you can flip back and forth. A small digital clock is always shown on the bottom of the screen.

5.3. Customizing the main menu

Notice that it is possible to select your own "favorites" to appear on the first page of the main menu. This is done for added convenience; if you place your most frequently used modules here (up to five), you will not have to browse through the entire menu too often. After you press "set menu favorites," you can select any of the 5 positions on the screen to place a module.
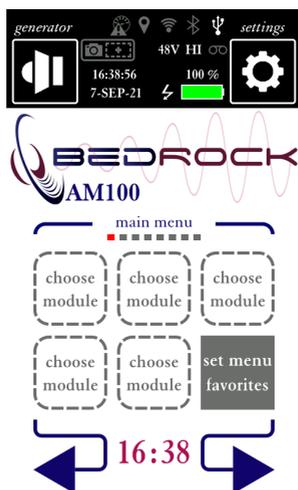
Appearance of the main menu page after pressing "set menu favorites."

After selecting a position (tap "choose module" in the location you wish to place a module), you can browse through the available modules. To clear a position where your previously placed a module, select the "choose module" placeholder from the list. You can place, remove and change modules in this menu screen as often as you like.

Some Bedrock distributors may offer specific modules that are not part of the standard set of AM100 modules (e.g. to cater to specific national standards). Such modules will then be pre-installed on the first page of the main menu, taking one or more of the slots otherwise used for your favorites. You can remove those modules from the first page of the menu, but since these modules are not listed on any of the other menu pages, you will need to put them back there before they can be used again.

Apart from the modules listed here in the main menu, the AM100 has specific operating modes triggered by the USB port: charging, data transfer and remote control. All modules and USB operating modes will be explained in detail in separate sections of this manual.

5.4. Top bar

At the top of the screen, you find a region with buttons and status icons ("top bar") that looks exactly the same in any module on the device. The top bar shows the status of all external connections (USB, Bluetooth, Wifi), the battery status, current date and time, and the status of the measuring system. The top bar also has two buttons: one to trigger a drop-down menu for the most important system settings, and another to access a menu for the signal generator and headset control.

The symbols, icons and buttons as seen in the top bar are explained below

generator	*Generator menu button* Drops down the generator / headset menu
settings	*Settings menu button* Drops down the settings menu
	Screenshot button Press to save a copy of the display as a PNG file to the internal-storage. The screen will freeze for a few seconds. The screenshot will be copied to flash as soon as the current measurement completes (a dialogue window appears).
14:35:49 17-JUN-21	*Current time and date.* Adjustable from the settings module (not the dropdown settings menu)
98 % ⚡	*Charge indicator.* The lightning bolt indicates a USB power connection. If the lightning bolt turns yellow, the battery is charging. If it turns magenta, the AM100 is powering a USB accessory.
●	Measurement running indicator. A red dot indicates that the measurement system is running (acquiring audio).
48V	*Phantom power status* Grey: disabled White: enabled, will become active during measurement Yellow: active (48V present on pins)
LO	*Indicates measurement range of mic preamp (LO / HI).* Can be set from the settings menu and determine overload/underrange boundaries.

⦾⦾	*Status of audio recording during measurements.* Grey: disabled White: enabled, will record a measurement once started Yellow: active (currently recording audio)
⚠	*Digital signal processor overload warning.* Grey: no warning Orange: processor close to maximum load Red: processor has been running at maximum load at any point during the current measurement. Verify measurement results and reduce load (lower resolution, fewer channels).
⚲	*GPS status (optional module on some AM100 models)* Grey: GPS disabled Green: GPS location acquired and being logged Red: no GPS lock, poor coverage.
📶 ⁎ �	*Wifi, Bluetooth and USB status* Grey: no connection White: connected Orange/red: connection warnings or errors

5.5. Drop-down menus: signal generator and settings

The signal generator/headset menu determines whether you are using the black TRRS connector (see section 5.7) for an external headset, as a signal generator, or not at all. Options that can be controlled from this menu are:

- *Enable/disable audio output.* Disable the generator/headset audio output if not used to conserve power.
- *Choose between headphones or generator.* The same audio jack (the one furthest away from the microphone) is used for either the signal generator or to connect a headset/headphones. Choose how you wish to use this connection.
- *Signal generator.* Start or stop, control the output gain, select the type of signal (STI, noise, sine waves) and the specific signal (e.g. STIPA rev. 5, 1000Hz Sine wave).
- *Headphones.* Monitor or mute, control the gain, select the monitored channel (primary measurement microfoon or 2nd input channel, through the jack connector).

The settings drop down menu is gives access to often-used system control settings.

- *Connect/disconnect USB.* If connected, the internal storage is presented to the USB host as a flashdrive. The device cannot be used for measurement in this mode.
- *Connect/configure WiFi.* Similar to USB connect, but in this case the device can be accessed wirelessly through WiFI.
- *Connect/pair BlueTooth.* Bluetooth is used for wireless remote control of the device.
- *Screen brightness.* Adjust to find a balance between comfortable viewing and power conservation.

- *Microphone range (HI/LO).* Choose the optimal range depending on the signal.
- *Enable/disable phantom power.* Disable when connecting directly to devices other than the microphone (not expecting 48V phantom power).
- *Enable/disable audio recording.* Enable to record the audio corresponding to each measurement (stored as a .WAV file with the same measurement number).

WiFi, USB and Bluetooth are used to transfer data to/from the AM100, to remote control the device, and for firmware updates. For information on how to establish and use a connection for these purposes, please consult the corresponding chapters of this manual (chapter 13-18).

Some settings cannot be changed while a measurement is running. The corresponding buttons will be disabled during measurements, but otherwise the drop-down menus can be used at any time, including while a measurement is running.

5.6. General layout of the user interface

The various measurement modules share a common user interface: most buttons and captions can be found in the same location throughout all modules. The graph below explains the most common icons and buttons.

Navigate through multiple screens using the white arrows

Start / Stop measrurement

Save measurement results to internal flash storage

Measurement ID number

Elapsed measurement time

Progress bar - turns green as measurement completes

Home button - return to the main menu

Area for module-specific data, settings and figures

Add text or voice annotation with measurement

Modules differ in terms of overall complexity, number of screens that can be displayed, and how elaborate the settings for the module are. However, the user interface is very similar between modules.

The buttons "start," "stop," and "save" are self-explanatory. Save commits all data from the current measurement to the internal flash drive on the AM00.

The last screen in any module is always the "browse measurements" screen. Here you can browse through the data from measurements previously saved on the device. Saved data can also be transferred to other devices and computers (see section 5.7 and chapter 13).

If the "notes" button is pressed, a screen appears that allows the user to type short text notes and/or record voice annotations.

Text notes are saved in a text file (.TXT), voice notes in a .WAV file - each with the same serial number as the current measurement. Voice annotations can be recorded either through the measurement microphone or a headset microphone (connected through the headset jack on the side of the device). The input channel can be selected in the "Settings" module.

The audio quality of the voice note recording is insufficient for measurement purposes. Voice notes are intended to record short reminders or descriptions of the measurement setup. Measurement audio can also be recorded, but this is a separate and independent option.

5.7. Copy measurements results to a computer

Browsing through saved data on the AM100 is not always the most efficient way to examine measurement results. It makes sense to copy data to a PC after every measurement session. A quick and easy way to copy your data is by connecting to a PC or Mac via USB. The AM100 then starts behaving as a flashdrive, allowing you to copy files from the device to your computer.

Results files on the AM100 are organized in separate folders for each of the modules (e.g. STIPA, RTA). Each measurement is stored as a separate .CSV file. In order to compile all of your data and assist in report generation, a series of MS Excel templates is provided (found on the flashdrive bundled with your AM100 and also available for download from www.bedrock-audio.com/support). Chapter 13 of this manual provides more information on how to work with saved measurement files.

5.8. Connecting microphones, headsets and other devices

The following connectors are found on the AM100:

XLR pinout
Pin 1. GND
Pin 2. Balanced +
Pin 3: Balanced -
Shield: Shield

Top of the device: XLR connector.

The XLR connector is used for to the primary microphone, either directly or through a standard XLR extension cable. Fit the silicone diffuser ring (provided) if you are not using an extension cable and your measurements are particularly sensitive to reflections against the top of the device. Note that 48V phantom power may be present while measuring (unless phantom power has been disabled through the settings menu).

2nd channel pinout (red)	**Headset pinout (black)**	**Generator pinout (black)**
Tip: Balanced +	Tip: Headphone right	Tip: Balanced +
Ring1: Balanced -	Ring1: Headphone left	Ring1: Balanced -
Ring2: GND	Ring2: GND	Ring2: GND
Sleeve: Shield	Sleeve: Microphone	Sleeve: do not connect
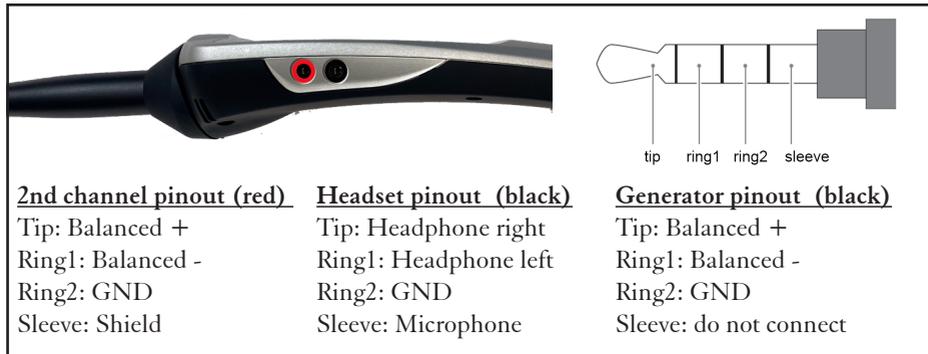
Left side: two TRRS connectors.

The front connector (red) is the second input channel. The rear connector (black) is used to connect a headset or as a balanced signal generator output. The headset connector is used to monitor the measurement microphone. If the headset features a microphone, this can be used to record voice annotations.

Right side: USB-C connector and status LED

The USB-C connector is used for charging, data transfer and connecting accessories. The color of the LED indicates its status:

Green	normal operation
Green/yellow blinking	charging
Red	operating system failure
White	hardware failure
Blue	wireless connection active (BT/WiFi)
Magenta	USB-C port is powering an acccesory

5.9. Some more tips and tricks

Add text or voice annotations to your measurements. Measurements made by the AM100 are uniquely identified by a serial number. However, as time passes by you may have a hard time remembering which measurement #000422 was exactly. A short text such as "1st floor, office 103" may jog the memory. If you press the button with the callout-icon marked "note" on the bottom right of the screen, a keyboard will pop up. Enter any text note that you like, or press the record button to record a voice annotation through the headset. The note and voice recording will be stored alongside the current measurement.

Record the audio of all of your measurements. Also from the settings menu, you can set the device to record audio (in .wav format) for all of your measurements. These .wav files are calibrated so that full scale corresponds to 150 dB SPL. You can play them back with any audio player, although they may have to be amplified to be heard at a satisfactory level (Windows is not set up to expect 32 bit audio files).

Back up your data via USB. All of your measurement data (and recorded audio) can be copied to a PC using the supplied USB cable. The AM100 simply presents itself as a mass storage device; see chapter 11 of this manual for the details. A word of warning: if your computer is set up to display hidden files, be careful not to change, move or delete any files from the SYSTEM folder.

Use a USB power bank for lengthy measuring sessions. Depending on the type and intensity of use, most users get between 6 and 8 hours of use out of a single battery charge. The battery is designed to last through the day during a typical measuring campaign. The battery is drained most quickly during the actual measurements, when all pre-amplifiers and the phantom power are fired up. When measuring continuously, expect the battery to last 3 to 3.5 hours. If this is not enough, we recommend using a (generic) USB power bank, commonly used for charging smartphones, connected via the USB port of the device.

Explore the settings module. Many options can be turned on from there, such as the option to delay the start of measurements, or to measure continuously until stopped.

6. Speech intelligibility measurements

6.1 Quick STIPA

The Quick STIPA module is intended to do simple and straightforward STI measurements using the STIPA test signal. A comprehensive explanation of the STI method is beyond the scope of this device manual; there are several online and offline resources that provide guidance and information on doing STI measurements, such as our own white paper on this topic.

In this section of the manual, we will briefly explain how to set up your equipment for measuring the STI. The Quick STIPA module makes the process as simple as possible.

For any STI measurement, you need a source of the STIPA test signal as well as an STI analyzer. In your case, the analyzer is your Bedrock AM100. You can choose whatever type of STIPA signal source that suits your application best (THe signal generator output of the AM100 itself, a TalkBox, a media player or otherwise).

The Embedded Acoustics reference STIPA test signal is included with the AM100 and can also be downloaded from the Bedrock website, free of charge. We supply IEC-60268-16 rev.4 and rev.5 test signals, fully compliany and compatible with all STIPA meters.
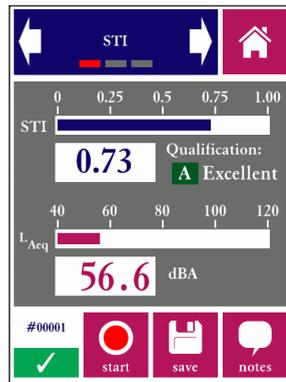
Measurement screen of the Quick STIPA module, in digit mode (left) and bar mode (right)

In order to carry out an STI measurement, complete the following steps:

- Power on your Bedrock AM100 and go to Quick STIPA
- Now start playback of the STIPA test signal through the channel or system which you intend to test.

- Note that the STI that you will now measure, will characterize all parts of your transmission path, from playback device up to the microphone of your Bedrock AM100.
- Tap "Start"
- Wait for 18 seconds while the measurement takes place. Make sure that signal playback is not interrupted, and that the acoustic environment is stable and free from impulsive sounds (such as door slams). *Note: the STIPA pro module uses a longer measurement time of 25s. A measuring time of 18s is a reasonable minimum, within the range allowed by the standard, to arrive at accurate results.*
- You will see that the STI on your screen stabilizing towards its end value.
- Optionally, you can press "save" ( ) once the measurement is finished to store the data, to be browsed or downloaded to PC later. A measurement number is assigned to each saved measurement. This number is displayed at the bottom left of the screen (preceded by a "#" character), above the battery indicator.

The steps described above are essentially the entire procedure. If you are new to STIPA measurements, please consult tutorials and standards to make informed decisions about the STIPA signal playback level, measuring positions, number of repeated measurements, etc.

The Quick STIPA module also shows some additional measurement information:

- The A-weighted sound pressure level
- The qualification band (a letter between A and U) that is used in some standards and defined in IEC 60268-16 rev. 4.
- A label ("bad"-"excellent") that characterizes intelligibility based on the STI

The arrows in the blue bar on the top of the display can be used to navigate through the different screens for each module. The Quick STIPA module has three screens: "STI", "settings", and "browse measurements".

The "settings" screen gives some options to control the appearance of your measurement results:

- Choose between presentation of the STI in large digits or as a bar plot
- Choose whether or not you want to see the qualification bands displayed

The "browse measurements" screen allows you to review earlier (saved) measurements without having to download your data to a PC first.

- Use the blue buttons at the bottom of the screen to scroll back and forth through your saved measurements.
- Note that the measurement number displayed above the battery indicator corresponds to the measurement currently shown on the screen.
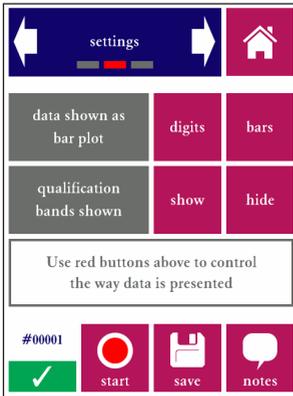
"Settings" screen and "browse measurements" screen. The latter screen is nearly identical to the STI measurement screen, but with previous/back buttons instead of start/stop and save buttons.

Reviewing larger numbers of STIPA measurements by using the "browse measurements" screen will be inefficient; we recommend downloading the data to a PC for this. Please refer to the section "accessing saved measurements through a PC."

Please take note of the following features of the Quick STIPA module:

- In the top bar, a red "recording dot" (●) is displayed whenever a measurement is running. At the bottom left of the screen, the progress of the current measurement is indicated by a progress bar (▬▬).
- The "save" button is only operational when new (unsaved) measurement data is available from a finished measurement. If this is not the case, the save icon is colored gray (🖫).
- The qualification band (a letter between A and U) is just a different way to represent the STI number; it is fully redundant. We currently recommend working only with the STI value itself, unless the use of qualification bands is explicitly required.
- The Bedrock AM100 calculates a heuristic reliability metric for each STI measurement. If the measurement is found to be unreliable (e.g. due to disturbance by impulsive sounds), then a red cross is displayed across the STI value. Crossed-out STI values are more often seen at the beginning of measurements, when the calculation has not ran long enough to compute a reliable STI. As soon as the cross disappears, the STI value can be trusted to be within the usual STI measurement error of 0.03.

A crossed-out STI value means that the measurement is not (yet) sufficiently reliable

6.2 STI average

This module is designed to do multiple STIPA measurements consecutively and take a (moving) average across up to five measured STI values.
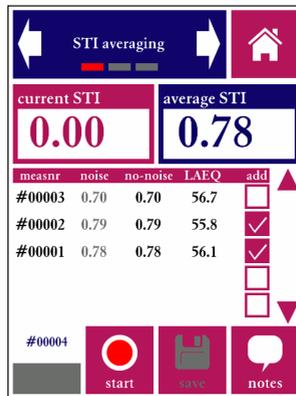
Place checkmarks in the STI values to include in the calculated average

Two STI values are displayed: the current measurement, and the mean across all measurements for which a checkmark is placed (up to 5 at a time). E.g., by checking the topmost 3 boxes, the average is calculated across the 3 most recent measurements.

For a further description of the functionality of this module, refer to the "STIPA Pro" section below.

6.3 STIPA pro

The STIPA pro module is also used to measure the Speech Transmission Index, but unlike the quick STIPA module, it also comprises features that allow you to analyze and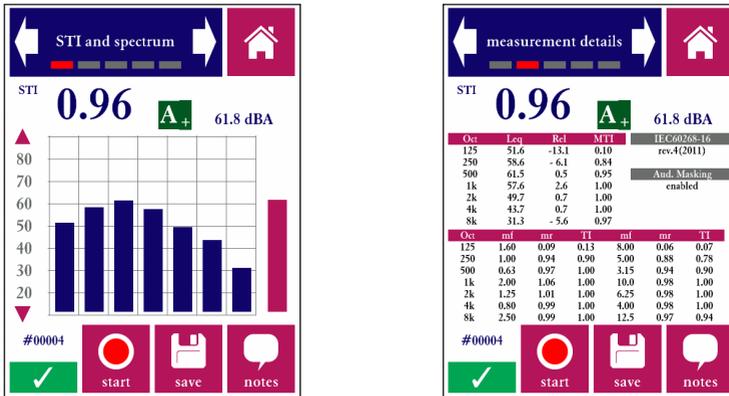 manipulate your measurement data in greater detail. It also uses a somewhat longer measurement time (25s instead of 18s for Quick STIPA), for improved measurement accuracy.

"STI and spectrum" and "measurement details" screen in the STIPA pro module

The STIPA pro module has five screens:

>**STI and spectrum**: display measured STI, octave spectrum and dBA level
>**Measurement details:** display all measurement details including the MTF
>**Settings**: configure the STIPA pro module.
>**Additive noise:** enter and enable/disable a noise spectrum to add to data
>**Browse measurements:** review saved measurements

Measurements are usually started from the "STI and spectrum" or the "measurement details" screen. STI measurements are set up and started in the same way as when using the Quick STIPA module (see above). However, in the STIPA pro module, more details about the measurement are shown, and the user has more elaborate options to configure the measurement. Measurements are started, stopped and saved in the same way as with the Quick STIPA module (by using the buttons at the bottom of the screen).

The "STI and spectrum" screen shows not only the current STI value and the A-weighted sound pressure level, but it also shows an octave band spectrum of the measured signal in the 7 octave bands from 125 Hz to 8 kHz. The STI analysis is carried out in these octave bands. The scale of the octave band spectrum plot can be modified by tapping on the triangular arrows to the left of the plot.

The "measurement details" screen shows all the diagnostic details of an individual STI measurement:

- OCT: The octave center frequency (Hz)
- REL: The octave band levels relative to the STIPA signal spectrum (this is effectively the frequency transfer function of the tested channel in octaves).
- MTI: Modulation Transfer Index. This quantifies the contribution to the overall STI from each octave band.
- mr: the m-values (modulation transfer function in octave band OCT for modulation frequency mf). These m-values are uncorrected, not adjusted for masking effects and additive noise. Since the STIPA signal features two modulation frequencies per octave band, there are also two m-values per octave band. The modulation frequencies are also shown (mf)
- TI: the Transmission Index per octave band for each modulation frequency

These measurement details will enable more experienced operators to determine not only the speech intelligibility of the tested channel, but also the *causes* of intelligibility reduction induced by the channel.
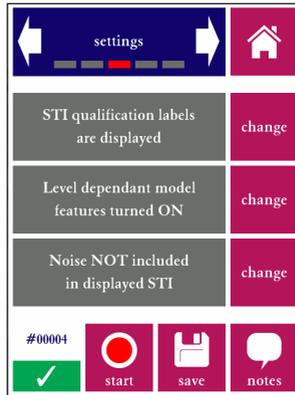
Settings screen

The settings screen gives the user three options:

- Choose whether or not to display STI qualification bands (A - U) as standardized in IEC-60268-16. We recommend using these only if it is explicitly required.
- Choose to enable or disable the level dependent model features: level-dependent masking and the speech reception threshold. *For standard measurements, these features should be turned ON.* Only when all-electric transmission chains are tested (without any acoustics involved) should these features be turned off, since there is no acoustic level reference in those cases.
- Choose whether or not to (computationally) add noise to the measurement.

There are two ways to include the influence of background noise in your STI measurement:

- Simply do your measurement in the actual noise environment. The STI method is designed to incorporate the effects of any noise present during the measurement representatively in the measured STI. This approach works well if the noise field is stable and free from fluctuating and impulsive components.
- If you have the option to physically "turn the noise off," then it is often more accurate to add the noise computationally in a so-called post-hoc calculation. The SM50 does this for you, if you enter the noise spectrum in the "additive noise" screen and change the setting to "Noise included in displayed STI."

If you wish to add noise computationally, the screen "additive noise" should be used to enter the noise spectrum. Note that the noise spectrum that you enter is only used if you also select the corresponding option in the settings menu: entering the spectrum is not enough, you also need to turn the additive noise feature on via the settings screen.
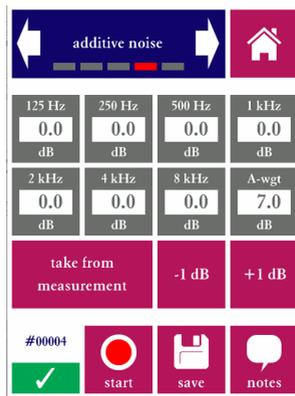
Additive noise screen

A few remarks about adding noise computationally:

- If you add noise computationally, all displayed STI values will be affected until you turn the additive noise feature off again. The main display field will now show the STI *including the effect of noise.*
- At the same time, the noise-free STI (as directly measured) is shown beside the indication "no noise:"
- Whenever additive noise feature is enabled, the label "N!" is shown next to the main STI display field. This is just a reminder that the displayed value includes the effects of (computationally added) noise.

- To enter a specific noise spectrum, simply tap each octave band field on the "additive noise" screen and enter the corresponding octave band level
- Alternatively, you can do a measurement (without STIPA signal playing!) of the current background noise. Although the STI value will be meaningless, the noise spectrum will be measured accurately. Save the measurement, and then press "take from measurement" () on the "additive noise" screen. When the keyboard appears, enter the measurement number for your background noise measurement.
- If you enter an A-weighted value after pressing the "A-wgt" field, all bands will be increased with an equal amount until the A-weighted level is matched. This is a quick way to lower or increase the overall level, and also to generate a pink noise spectrum of a given A-weighted level (as the default spectrum shown is pink).
- The buttons "-1 dB" and "+1 dB" adjust the levels in all bands by 1 dB.

You can browse through all measurements saved earlier in the "browse measurements" screen. The following tips and pointers may be useful to remember when browsing STI measurements with Quick STIPA and STIPA Pro:

- Measurements saved in Quick STIPA can be browsed in STIPA pro, and vice versa. The measurement numbering range is shared between these modules. This means that you can use STIPA pro to investigate measurements done with Quick STIPA in more detail using the browsing screen in STIPA pro.
- All data for each measurement is always saved - including any data not actually shown during the measurement itself, including the time and date.
- Measurements can also be retrieved over USB; please refer to the section "accessing saved measurements on a PC" for further instructions. For larger numbers of measurements, processing of these saved data files (e.g. through worksheet software) may be more efficient than inspection through the measurement browser.
- If you wish to reset the measurement number (restarting at number 1), you need to connect the device to a PC. Remove all measurement files (.CSV) from the directory STIPA, as well as the file _INDEX.DAT.

6.4 Full STI

The Speech Transmission Index (STI) can be measured by means of various test methods. The STIPA method, as implemented by the Quick STIPA and STIPA pro modules, is a fast and efficient version of the STI originally developed for public address systems. Although fast and robust, STIPA has its limitations: only 2 modulation frequencies are measured per octave band, whereas the complete Speech Transmission Index measurement scheme calls for 14 modulations per octave band. This means that STIPA is undersampled in the modulation domain. For many systems, the measured STI is not (or hardly) affected. However, some cases call for "Full STI" measurements:

- Whenever echoes are present (delayed loudspeakers, large halls and cathedrals)
- When tested systems or channels feature complex combinations of distortion factors.

Full STI measurements were available in commercial measuring tools up to the late 1990s, but have subsequently been absent from the market for two decades. The fact that a full STI measurement used to take around 15 minutes made Full STI measurements entirely impractical. Also, the introduction of STIPA around 2002 made a reasonable alternative available.

The Full STI module now available on the AM100 fully implements every aspect of the Speech Transmission Index model, making it the most universally applicable speech intelligibility metric to date:

- Fully compliant with the Full STI definition according to IEC-60268-16 rev.5 (2020)
- Computes the complete 14x7 Modulation Transfer Function matrix
- Uses an efficient algorithm that completes the measurement in approx. 1 minute
- Implements the *direct measurement method* as defined in IEC-60268-16.

The "direct method" (in which a modulated test signal is presented to the channel under test), as implemented here, has crucial advantages over the "indirect method" (where the STI is estimated based on a measured impulse response). The indirect method implicitly assumes that the channel under test is a linear time-invariant system. Whenever electroacoustic components are involved, the tested channel never fully satisfies the linearity requirement. Indirect estimates are notoriously inaccurate in many real-life scenarios, but have nonetheless been used extensively, for lack of alternatives based on the direct method (such as this module).

The Full STI module implements a novel measurement algorithm, that leverages the advanced signal processing capabilities of the AM100. This algorithm, which is documented separately in an Embedded Acoustics white paper, has: he following characteristics:

- A novel test signal is used, which repeats 7 different patterns (segments) in a circular fashion.
- Each pattern roughly follows the approach of STIPA: each octave band is simultaneously modulated with 2 modulation frequencies.
- Each pattern takes approx. 9 seconds to complete. The whole measurement is completed in 65 seconds.
- The patterns are set up in such a way that all modulation frequencies (out of the set of 14) have been used in each octave band once the full 65 second cycle is completed. Hence, all 14 modulation frequencies are tested in all 7 octave bands.
- The first phase of a new measurement is a synchronization phase, during which the analysis algorithm determines when to expect transitions between patterns.

The Full STI test signal is included on the flashdrive bundled with the AM100 and also available on the Bedrock BTB65 Talkbox.. Note that the Full STI signal is copyrighted material; all audio files of the signal released by Bedrock Audio are watermarked. Use of the signal other than directly with the AM100 requires a license from Bedrock Audio. The AM100 can also generate the Full STI signal through its built-in signal generator.

The measurements themselves are no more difficult or complex to carry out than STIPA measurements. The procedure is largely the same; only the duration is longer and the results are more univerally representative of true speech intelligibility.
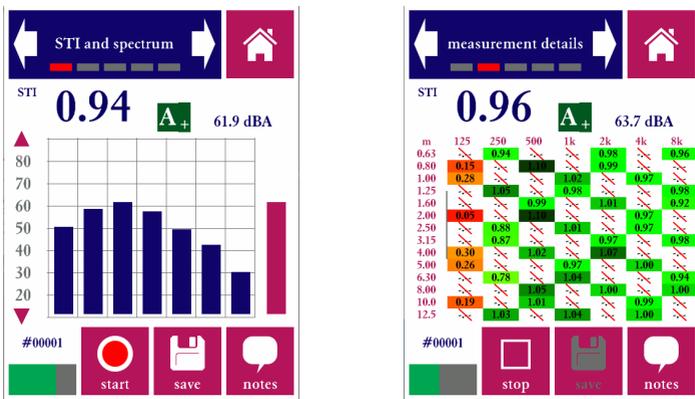
Full STI measurement result (left) and MTF details (right)

The Full STI module has the same overall design as the STIPA pro module, including the option to computationally add noise. The main difference is the second screen, which visually represents the Modulation Frequency Transfer (MTF) matrix as it gradually fills up during the measurement. The user has the option to stop the measurement before the full cycle has completed, shortening the duration of the measurement. This makes sense if the STI remains stable across multiple phases of the signal. The result of a prematurely

ended Full STI measurement can be thought of as a compromise in between STIPA and Full STI.

Stopping (and saving) a Full STI measurement prematurely, before the ma trix has been completely sampled, does not render the measurement invalid. However, the MTF matrix will be sparse, and the output file may contain a smaller number of data fields than expect by Excel tools and third-party software.

Please refer to the section on the STIPA pro module (6.3) for a description of the various features and controls of the module (additive noise, qualifications bands, disabling level-dependent masking, etc).

6.5 Speech Level Meter

In the context of speech intelligibility measurements, special care is needed when determining the sound pressure level of running speech. As human talkers use their articulators to form speech utterances, they alternately produce high-intensity bursts of acoustic energy and naturally occurring pauses. Across the length of a spoken sentence, periods of silence roughly account for half of the total time (length) of the sentence. Those pauses do not necessarily contribute to the intelligibility of the sentence, but they do have an influence on the overall (equivalent continuous) sound pressure level of the sentence. There is a firmly established relation between speech-to-noise ratio (SNR) and speech intelligibility - but how do we determine the speech level to calculate (SNR)?

It has been shown that removing the pauses from speech increases the correlation between SNR and subjective speech intelligibility. In the 1990s, a method was developed within European ESPRIT project 2589 (SAM) to assess speech levels while removing the influence of natural pauses. This method is known as Speech Level Meter (SLM) and is included in IEC-60268-16 as part of a normative annex (Annex J). This Speech Level Meter module is a direct implementation of this method.

The procedure consists of the compilation of a level histogram, which normally reveals a bimodal distribution: a low-level mode which represents the background noise, and a higher-level mode which represents speech. The RMS of speech and background noise can be calculated separately by summing the two modes in the histogram separately. The criterion for separating the two modes is a threshold that is set 14 dB below the SPL presumed to correspond to speech.

Estimated speech level (left) and histogram on which the estimate is based (right)

To use the Speech Level Meter, simply carry out a measurement while sampling running speech. The SPL of speech and background noise are calculated, as well as the total LAeq.

7. Level and spectrum

7.1 SPL meter SPL meter

The Bedrock AM100 as a Sound Pressure Level meter (also known as SPL meter or SLM) complies with IEC 61672 Class 1 and ANSI S1.4 Type 1. It is classified as a group X instrument.

The SPL meter modules features three screens: SPL measurements, settings and browse measurements.
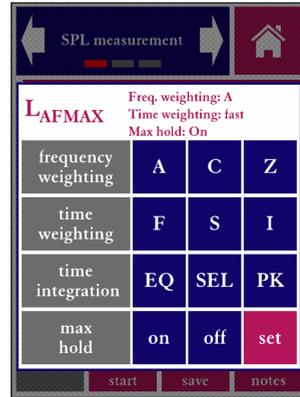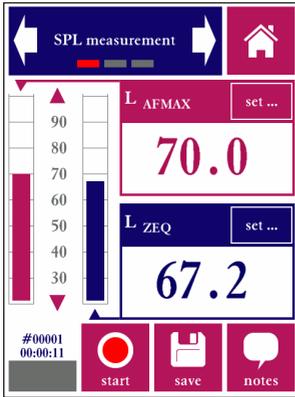
SPL measurements screen, during measurement (left) and with settings keyboard showing (right)

The following sound pressure level measurement settings are supported:

- Frequency weighting: Z (linear, unweighted), A-weighted, C-weighted
- Time weighting: Fast (F), Slow (S) and Impulse (I)
- Equivalent continuous (EQ): computes the time-integrated level over the measurement interval; the level shown corresponds to the level of a continuous signal containing the same amount of energy as the measured signal.
- Sound Exposure Level (SEL): time-integrated value as described under EQ, but recalculated but normalized to 1 second.
- Peak values (PK): the highest instantaneous value within an interval
- Max hold (holds and displays the maximum level during the measurement interval; used with Fast, Slow or Impulse time weighting)

Note that MAX and PK are not the same; MAX represents the highest measurement over a short time averaging period (either FAST or SLOW), and PK represents the highest instantaneous value in the signal (even if this is a peak of very short duration).

The AM100 allows PK measurements in combination with any frequency weighting, but only LCPK is commonly used.

Once a measurement has started, each combination of the above settings is measured simultaneously. However, not all of these combinations are displayed: only two level measurements are displayed at the same time. Each of these two levels is shown in its own bar graph and its own digit field. The red graph corresponds to the red field, the blue bar graph to the blue field. You are free to select which level to display in each of the two fields. Tap the "set" button in the level field to choose time and frequency weightings.

The AM100 will warn you if the signal level is outside the linear range of the microphone. This occurs if the signal level is too low in comparison to the noise floor of the microphone (underrange ⚠) or if it approaches the maximum acoustic level that the microphone can handle (overload ⚠). Both types of range warnings are shown as a warning label next to the corresponding measurement value.
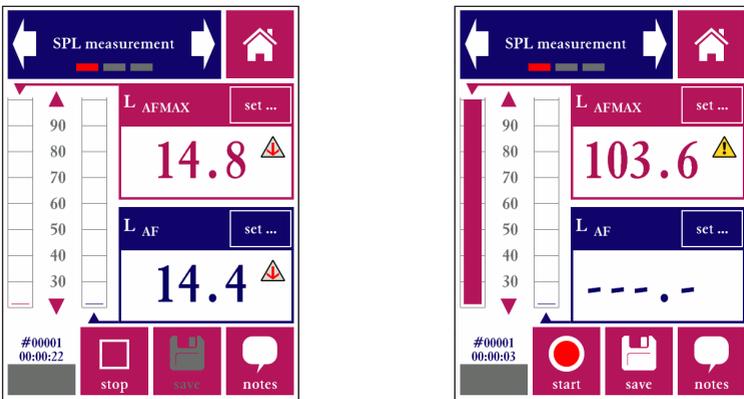
Range warnings are shown if the signal level is either too low or too high

The limiting factor for the overall dynamic range of the measuring system is usually the microphone. If the standard microphone is replaced by a microphone of a different type, the displayed overload and underrange warnings may not be reliable. Please check the Bedrock audio online support pages to find out for which microphone types (besides the standard microphone) range warnings are supported. If you replace the microphone, it may be necessary to patch the system firmware with a specific update package.

For MAX, EQ, SEL and PK measurements, the range warnings will be displayed from the moment the underrange or overload condition occurs until the measurement is restarted. For FAST and SLOW measurements, the range warning will disappear once the overload/ underrange condition has passed.

The settings screen in the SPL module allows you to set the measurement time. Once you press start, the measurement will run for as many seconds as you choose here. You may also choose to let measurements run continuously; you then determine the measurement time manually, by choosing the moment you press "stop."

The SPL meter module also features are measurement browser, that lets you review previously saved sound pressure level measurements. Note that the measurement browser will not display FAST or SLOW values, only MAX, EQ and PK. The reason for this is that FAST and SLOW values are fluctuating measures; these are not single values characterizing the measuring interval corresponding to any measurement number, and therefore cannot be displayed except in real-time, during the measurement itself.

A few things to note about SPL measurements and the SPL meter module:

- By definition, "max hold", "EQ/SEL" and "PK" are mutually exclusive.
- The elapsed measurement time is shown above the progress bar. If the measurement interval is set to "continuous," the measurement will run until it is manually stopped. Note that, in practice, the maximum measuring time is limited by:
  - The battery. When measuring continuously, the battery is drained relatively quickly (within 3-4). For longer measurements, we recommend using the charger or an external power bank.
  - Internal data storage, especially if you are recording all measurement audio (per the option in the "Settings" module). When recording audio, the internal storage fills up in about 5 hours.
- When you press "save," data from the current measurement is stored. Results based on all time and frequency weightings are saved; not just the results that are currently displayed. However, as remarked above, since the instantaneous levels fluctuate throughout the measurement, these cannot be saved as a single number representing the entire measurement interval. So for each saved measurement, just the maximum level (max hold) and equivalent continuous level (EQ) and Peak level (PK) are saved.
- As a logical consequence, the measurement browser can only display Max Hold, EQ and PK results from earlier measurements. If you set a display field to show instantaneous values, the measurement browser will leave this display field blank.
- You can use the measurement browser to go back to earlier measurements, and see what the level is with different time- and frequency weightings applied.
- The measurement numbers form a separate consecutive series for each module. That means that the same measurement number might occur multiple times (once for each module; e.g. number #0002 might exist for SPL as well as RTA, STIPA, etc.)

7.2. Real-Time Analyzer (RTA) RTA

The RTA module is intended for frequency analysis in the range from 20 Hz to 20 kHz. The bandwidth of the frequency bands can be set to 1/1, 1/3, 1/6 or 1/12 octaves. We recommend 1/1 or 1/3 octave analysis for most uses, unless a finer frequency resolution is needed.

The operation is generally similar to the SPL module, but instead of just the overall signal level, a frequency analysis (into separate frequency bands according to IEC-61260:2014 class 1) is also shown.
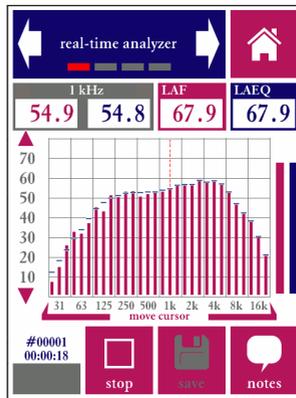
RTA main measurement screen (1/3 octave band mode)

The RTA module has the ability to present two different spectral views at the same time, differing in frequency weighting (A, C or Z) and time averaging and integration (FAST, SLOW, IMPULSE, MAX, or EQ). The main spectrum is presented in red bars, while a second (auxiliary) spectrum is shown in blue lines. The corresponding broadband levels are also shown, to the right of the spectrum.

While the spectrum graph already gives a first impression of the spectral content of the measured signal, you may wish to read the exact value in each band. This is done by placing the cursor (dashed line) over the band of interest. The cursor can be moved left and right by pressing on the left or right side of the frequency axis (or the area of the graph above the axis). The values for the selected band are shown in the grey field on the top left of the screen (main weighting shown in read, aux weighting in blue - same color as the spectrum bars).

The "measuring time" screen can be used to set the duration of a measurement to a fixed value. The measurement can also be set to run until stopped manually. The "settings" screen allows you to switch frequency resolution, and to change the weightings for the main and auxiliary spectrum views.

By pressing the "save" (🖫) button in the main measurement screen, you store the current measurement on the devices internal storage, using the measurement number indicated above the battery indicator.

Some remarks about the RTA module:

- By definition, "max hold" and "EQ" are mutually exclusive settings.
- The elapsed measurement time is shown above the progress bar. If the measurement interval is set to "continuous," the measurement will run until it is manually stopped. The progress bar itself is not used when measuring continuously.
- Once saved, measurements can only be retrieved with the same frequency resolution (i.e., a measurement carried out with 1/1 octaves cannot be retrieved as an 1/3 or 1/6 octave band spectrum).
- However, data recalled through the measurement browser are shown with the frequency weightings selected at the time the measurement is recalled - even if these are different from the settings that were in effect during the measurement. This allows you to inspect the effect of frequency weightings on all measurements saved previously
- Note that the spectrum and its corresponding broadband level are affected by the same time- and frequency weightings. In other words, if the main spectrum is A-weighted, then the level shown in the red field is also A-weighed (and vice versa). If you wish to display a Z-weighted spectrum and an A-weighted level at the same time, you need to display these in different channels (main and aux).
- The measurement numbers form a separate consecutive series for each module. That means that the same measurement number might occur multiple times (once for each module; e.g. number #0002 might exist for SPL as well as RTA, STIPA, etc.)
- The filters comply with the requirements of IEC-61260-1:2014 class 1 and IEC-61260:1995 class 0.

7.3. Fast Fourier Transform analyzer `FFT`

The FFT module allows you to perform spectral analyses with a much higher frequency resolution that the RTA module. This is very useful if you are looking to identify the exact frequency and level of certain signal components, or if you want to study the harmonic structure of sounds.

The maximum frequency resolution of the FFT analysis, with supported window sizes of up to 32768 samples, approaches 1 Hz. This means that the analysis contains more data and more detail than can be shown on the display at once. In order to deal with this, the FFT module features a zoom function to zoom in on part of the spectrum. Also, the full details of every measurement can be retrieved through USB by saving measurements.
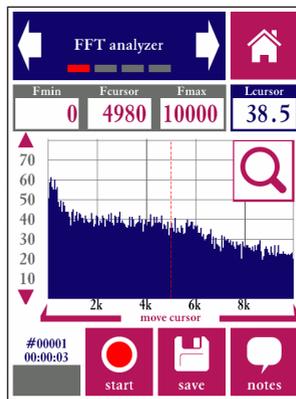
FFT measurement screen

The general "look and feel" of the measurement screen is similar to the RTA module, but with a few differences. First of all, in line with the normal conventions, frequency weightings are not supported by the FFT module. Secondly, a button $\boxed{Q}$ at the top right-hand side of the graph can be tapped to reveal a small keyboard with zooming options:

These controls are used to zoom in or zoom out along the frequency axis, and to center the display around the current cursor position. By pressing "hide" the zoom controls are collapsed again. The zoom controls are used together with the cursor to focus on any region of interest in the spectrum.. The cursor, shown as a vertical red line, can be moved by tapping the arrows below the frequency axis or by tapping directly on the graph.

The minimum and maximum frequency that can currently be displayed are indicated at the top of the screen, as well as the exact frequency at the cursor position and the level at the cursor position.

The second screen of the FFT module allows you to set an integration time for the FFT analysis. If the integration time is set to "infinite," then the instantaneous spectrum is shown. The graph on the display will show rapid fluctuations. In general, the longer the integration time that is chosen, the more accurate the spectrum.

The third screen of the FFT module ("settings") is where the parameters of the FFT algorithm are controlled.

The window size determines the frequency resolution of the FFT analysis. The number of spectral lines is exactly half the window size. At the maximum window size (32768), the audible frequency range is analyzed in16384 spectral lines. Shorter windows offer less detail, but the analysis runs quicker allowing for shorter integration times to arrive at the same accuracy.

The type of window to be applied can also be set in the graphics screen. Each type of window (blackman, hamming, rectangular, etc) is associated with a specific set of advantages and disadvantages relating to the FFT analysis. We recommend studying the literature on spectral analysis, or one of the many available online resources, to choose the window that best suits your application.

For most non-critical applications, the settings selected by default (hamming window with a window size of 8192 samples) are a suitable choice.

Some remarks about the FFT module:

- The elapsed measurement time is shown above the progress bar. If the integration time is set to "continuous," the instantaneous spectrum will be shown.
- The FFT module is recommended whenever a high frequency resolution is needed. To obtain standardized spectral measurements, where the exact levels at standardized frequencies are of prime interest, 1/3 or 1/1 octave band measurements through the RTA module are the best choice.
- Saved data can be retrieved over USB and imported in MS Excel or any other program capable of dealing with .csv formatted data. This is the same as all other modules. However, the FFT module is capable of producing much more data (up to 32768 data points and frequencies for a single measurement). Not all spreadsheet programs may be able to import that many columns, which is why the data is organized in rows rather than columns.

7.4. Statistical acoustics (percentiles) L%

This module is designed to create insight into the statistical distribution of sound levels during the measurement period. In particular, it give percentile levels: the level exceeded during the measurement period for a specific percentage of the time. Percentile levels are commonly indicated with letter L (for level) followed by the percentile threshold, e.g. L95 being the level that is exceeded 95% of the time during the measurement period. and L10 the level that is exceeded only 10% of the time. Logic dictates the L95 normally has a lower value than L10; for a perfectly steady-state signal, L95 and L10 will both be equal to LEQ.

The L% module offers the same frequency weightings and time averaging options as the "standard" sound level meter module (A, C, Z; FAST, SLOW, IMPULSE).

⚠ The operation and specifications of sound level meters are very specifically and precisely determined by IEC-61672-1:2013. However, there is currently no international standard that governs how the additional parameters that come into play with percentile-levels should be chosen. Implementations by various manufacturers, while these appear to be consistent on the surface, may differ. The procedure adopted by the Bedrock AM100 is explicitly defined further on in this chapter.

The module is capable of simultaneously measuring and displaying sound pressure levels with six different sets of settings: three displayed on the left ("main") and three displayed on the right ("auxiliary").

Percentile settings (including Min, Max, EQ, Peak, Current) can be chosen by pressing the level field, in which case a keyboard appears
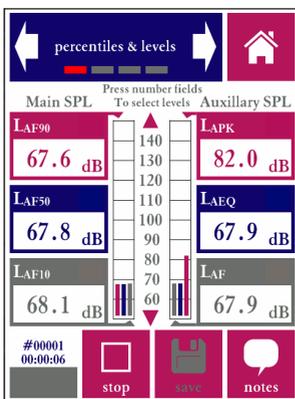
Statistical acoustics module, showing levels for six different sets of settings (left). The percentile selections can be chosen through a keyboard that appears when pressing the level fields (right).

The frequency weighting and time averaging options are selected in the third screen of the module, in the same fashion as the standard sound pressure level module. Two sets of time and frequency settings can be chosen (main and aux), respectively corresponding to the three fields on the left side and right side of the first screen. The measuring time is set on the second screen, in the same way as all other modules.

Note that each level measurement is defined by three different settings:
- Frequency weighing (A, C, Z)
- Time averaging (FAST, SLOW, IMPULSE)
- Statistic (MAX, MIN, CURR, PEAK, EQ, or any percentile between 1 and 99%)

> Not all implementations of percentile level measurements explicitly allow time averaging settings to be chosen. The time averaging window is then (often implicitly) set by the manufacturer of the device. The length of the time window may vary widely, from less than a millisecond to several seconds. If the standards and documentation that you are using to guide your measurements are unclear on the topic of time averaging, we recommend choosing FAST. The time averaging window is then set to 125 ms and opt for a measurement duration of 60 seconds or longer.

The percentile levels are computed according to the following procedure:

- A level measurement takes place across audio frames of 512 samples, at a sampling rate of 48 kHz. A computation takes place after the acquisition of every new audio frame (so 93.75 times per second).
- For each channel (Main and Aux), the frequency-weighted and time-averaged level is calculated and rounded to 0.1 dB.
- A level histogram is compiled with bins (categories) that are 0.1 dB wide. Each measurement is sorted into this histogram. I.e., 93.75 measurements per second are added to the histogram.
- The percentile levels are calculated from the histogram by determining which level constitutes the threshold which is exceeded more than the chosen percentile (e.g. for L95 more than 95% of the samples in the histogram exceed the threshold level).
- Max and Min are respectively the highest and lowest bin in the histogram for which data is recorded. EQ is the energetic average across all measurements. By selecting CURR, the current level is shown according to the selected time averaging option (FAST or SLOW).

8. Logging modules

8.1 LEQ logging

The LEQ logging module is intended for long-term monitoring and logging applications. While LAEQ logging is most common, the module also supports C and Z frequency weighting. When running this module, the Bedrock AM100 keeps track of how the equivalent-continuous sound pressure level (LEQ) develops over time. This is displayed on the device and also logged, for download to a PC later on. All frequency weightings (A,C,Z) are logged, but only one weighting is displayed. By default the displayed weighting is set to A. This module can be used for several purposes:

- monitoring of environmental noise sources
- monitoring and logging of sound levels at music venues
- measuring sound levels of multiple events across a large time span

Instantaneous and time-averaged data are shown on the display. If you enter a level limit, a warning ("traffic light") will be displayed to indicate whether the limit is currently being exceeded. The log file is automatically saved, and can be used to demonstrate that (legal) limits were not exceeded.

The LEQ logging module measures three different metrics:

- The LEQ during a short time period between 0.1s and 60s. This is level ("LAEQ current") can be thought of as the current contribution to the overall sound exposure.
- The LEQ across a longer time span (1-60 minutes), calculated as a movering average of a series over shorter time periods (i.e. a moving average of the series of "LAEQ current" measurements).
- The current instantaneous level with FAST time weighting (LAF). This measure is not logged; it is only displayed as in the form of a bar plot, to obtain a quick impression of how the instantaneous level at any moment is affecting the LAEQ.

The LAEQ module showing the moving average LAEQ across 15 minutes.

The LEQ logging module has a start/stop button similar to all other modules, but unlike other modules, it has no "save" button. Since this module is specifically intended for logging purposes, all data is always saved to a text file. Whenever a new measurement is started, the measurement number is automatically incremented and a new file is opened. This means that data is never inadvertently discarded (e.g. by forgetting to press save after a lengthy session).

⚠ LEQ logging sessions can run for a long time, up to many days. Before starting a long session, check if there is enough free storage on the device. When recording audio (an option available through the settings module), the memory fills up in less than 24 hours. We recommend disabling audio recording on long sessions.

⚠ The log files of sessions that span multiple days are split up. Every day at precisely midnight, the log file is closed and a new file is opened.

As an example (for the purposes of this manual), we will assume that time averaging takes place over a window of 15 minutes, resulting in LAEQ15MIN (which is a commonly used setting for sound exposure monitoring at music venues). However, note that the averaging time can be adjusted to any period between 1 and 60 minutes.

The LAEQ15MIN as displayed on the screen is the moving average of a series of "LAEQ current" measurements. The shorter the time period used for LAEQ current, the more values are used to compute LAEQ15MIN (and the longer your log file will become, since a line is written for every LAEQ current period). If you choose a short period for LAEQ current then you will be able to determine fluctuations in the measured level more accurately, but you may end up with a lot of data to sort through. There is a trade-off; the optimal choice depends on the application.
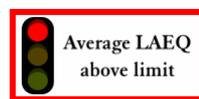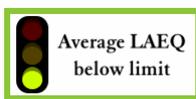
"Traffic light" indicating whether the limit is being exceeded.

If the LAEQ15MIN exceeds the preset level limit, the traffic light indicator will turn red. If the LAEQ15MIN is still below the limit, but LAEQ current exceeds the limit, this implies that the LAEQ15MIN will eventually reach the limit if the current levels do not go down. In that case, the traffic light will be yellow/orange. If the LAEQ15MIN as well as the current value are below the limit, then the traffic light shows green.

The first screen of the LEQ monitoring module (see previous page) gives the levels in large digits, and creates a quick overview. The second screen (see next page) shows a graph of the running series of "LEQ current" measurements, which gives a good feel of how the overall level is developing.
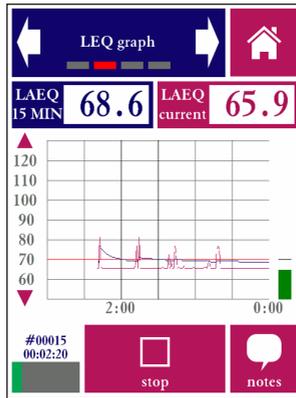
Screen 2 of the LEQ module, showing a graph of how the average LAEQ is developing.

In cases where the operator has control over the measured sound level (i.e. through control of a PA installation), and is using the Bedrock AM100 to make sure that the level is kept below the limit, the graph gives a useful insight into how much "room" there is below the limit. In cases where noise events (such as train passage) are monitored, the graph shows a recent history of events.
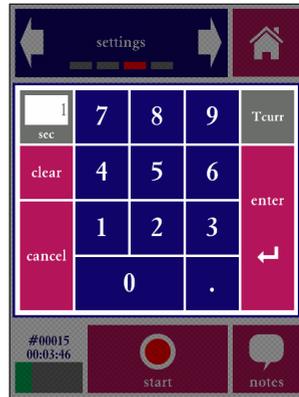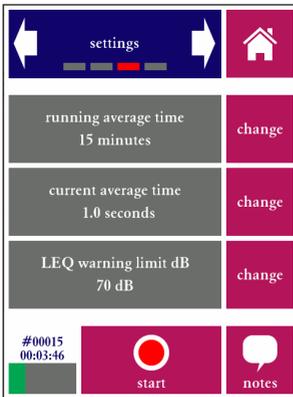
Settings screen; the right picture shows the keyboad that appears when the "change" button is pressed to adjust the warning limit

As with all modules, the LAEQ logging module has a screen which can be used to set all parameters (in this case: LAEQ averaging time (1-60 minutes), current intervals (0.1-999 seconds) and the level limit. Values are entered through a keyboard that appears upon pressing "Change."

Browsing through measurement data

Browsing through measurement data works a little differently than in other modules, since the amount of data associated with a measurement file can be very large. In many cases, it will not be practical to inspect the data from the device itself, and data will be downloaded to a PC first.

The blue buttons are used to browse through all earlier log files stored on the device. Note that each file may span anywhere between a few seconds and many hours, depending on how much time elapsed until the "stop" button was pressed. Within each file, you can scroll through the data with the red arrow buttons (single arrow to scroll line-by-line, double arrow to quickly scroll page-by-page). Each line shows the time, the current level for the corresponding measurement interval, the average LAEQ and an indication whether or not the limit was exceeded.

⚠ Long sessions may be broken into multiple files by pressing stop and (immediately) again start. If the device crashes unexpectedly (e.g. in case of power failure), there is a chance that (part) of the log file may be lost. By breaking the session up in multiple files, the risk of losing data is reduced.

⚠ For long sessions (>3 hours), you will need to power the device from an external power source (such as a PC, power bank or the USB charger). A new log file is automatically started once every 24 hours, at midnight exactly.

⚠ Note that the log file contains more information than shown on the device. In particular, not just the time but also the date is stored, and also C- and Z-weighted levels. Make sure that the date and time are set correctly (though the settings module).

8.2. RTA logging

The RTA logging module is intended for long-term monitoring of the spectral characteristics of sounds. It logs the spectrum in 1/1, 1/3, 1/6 or 1/12 octaves, as well as the A, C and Z-weighted equivalent continuous levels. The averaging window can be set to any value between 1 second and 999 seconds.
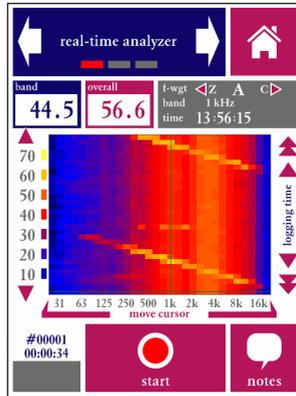
RTA main measurement screen (1/3 octave band, 1 second averaging)

On the main screen of the RTA logging module, logged data is represented in the form of a spectrogram. The horizontal axis is the spectral axis (bandwidth between 1/12 and 1/1 octave), the vertical axis is the time axis. The latest spectrum is added consecutively to the bottom of the screen. The measured level is indicated by means of color, according to the legend displayed on the left side of the screen.

The RTA logging module does not feature a "save" button. Measurement data is automatically logged to an output file after every averaging interval.

Once logging is started, the logging will continue indefinitely unless it is stopped again manually (through the "Stop" button) or until the battery runs out. For long term-logging, we recommend connecting the instrument to an external power supply.

A new log file is automatically started every day at midnight.

8.3. Environment & Safety

The E&S module logs a combination of metrics that is useful for long-term monitoring needs that arise from a variety of (inter)national standards. In short, the E&S module logs the RTA spectrum (1/1 or 1/3 octaves) as well as all combinations of time- and frequency weightings and percentile levels. Most of these measures can also be shown on the display while logging. Similar to the LEQ logging module, a logging graph of current and cumulative levels is also shown on the display.
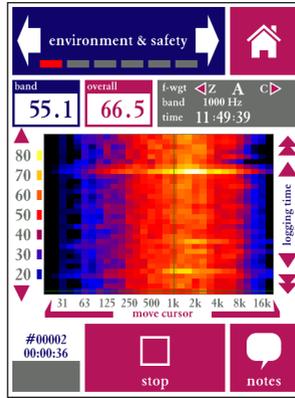
E&S module showing broadband levels and RTA logging

The E&S module is effectively a combination of multiple other modules. As a consequence, the log files can become quite large, especially when low logging intervals (down to 0.1s) are chosen.

The E&S module does not feature a "save" button. Measurement data is automatically logged to an output file after every averaging interval.

Once logging is started, the logging will continue indefinitely unless it is stopped again manually (through the "Stop" button) or until the battery runs out. For long term-logging, we recommend connecting the instrument to an external power supply.

A new log file is automatically started every day at midnight.

8.4. DIN15905-5

DIN15905-5 is a German standard for monitoring and logging sound levels at live music venues. The dedicated module named after this standard provides the exact measurement metrics and calculations as defined in this standard, leading to a level history graph similar to the LEQ logging module. However, two levels are logging (and displayed in a graph) simultaneously: LAEQ and LCPK.
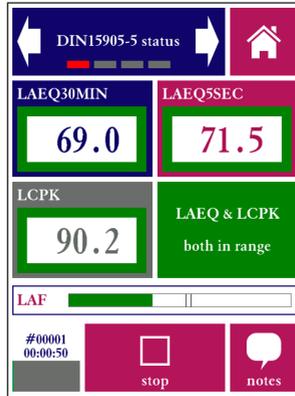
Screen that shows both logged metrics (LAEQ30MIN and LCPK) as well as LAEQ5SEC and LAF. The green field turns red if the limits are exceeded.

The limit for LAEQ30MIN is 99 dB(A), the limit for LCPK is 135 dB(C).

The levels are to be measured at the worst-case location where the public may be present (usually close to loudspeakers). This may not always be feasible; the standard allows for a correction method as follows:

- Take a measurement (before the event) at the relevant (worst-case) location (German: *massgeblichen Immissionsort*).
- Take a measurement at a reference location where the measurement microphone can stay during the event (German: *Ersatzimmissionsort*).
- Calculate correction factors for the A-weighted and C-weighted levels (resp. called k1 and k2) and apply these to the measurements.

The DIN15905 module features buttons to start both SPL reference measurements, from which k1 and k2 are automatically computed. Alternatively, it is possible to input k1 and k2 manually.

9. Room acoustics

9.1. Reverberation Time meter (RT60) RT60

The reverberation time (or RT60) is among the most commonly specified acoustic measures. Unfortunately, reverberation time measurements are also among the most complex to carry out accurately. RT60 measurements are prone to errors, which cannot always be detected by the measuring device.

> ⚠️ If you are new to measuring reverberation times, we recommend that you read this section of the manual carefully, and preferably also consult external literature and training materials on reverberation time measurements.

Standardized procedures for measuring the reverberation time of a room are specified in ISO-3382-2. Generally speaking, there are two different ways to measure RT60: the impulse response method and the interrupted noise method. Both methods are standardized through ISO-3382-2. The impulse response method uses a carefully controlled, known test stimulus of short duration (sweeps, chirps, pseudo-noise sequences or gunshot-like impulsive sounds). By normalizing the recorded sound in a room relative to the source signal, the impulse response of the room is obtained. From this impulse response, the RT60 is calculated.

Starting with firmware version 2.4.0, the impulse response method (which previously not supported) has also been implemented on the AM100. The test signal to be used is **a loud, short impulsive sound** (such as the sound from a starter gun). Do *not* use sweeps or chirps as your test stimulus.

> ⚠️ If you are using impulsive noise as your test stimulus, even the slightest movement of the microphone during the test may render the measurement inaccurate. The use of a tripod is absolutely necessary.

The interrupted noise method, which is also supported by the Bedrock AM100, measures the decay of the sound field if a sound source is suddenly switched off. This method is more robust against slight inaccuracies or inconsistencies in the placement of sound source and microphone. In small rooms, no other tools than the Bedrock BTB65 talkbox and the AM100 may be needed. In larger rooms, a larger (spherical) loudspeaker and amplifier will be needed. For the interrupted noise method, the following steps are taken in preparation:

- Place a source of continuous sound (preferably pink noise) in the room for which the RT60 needs to be measured.
- Place the Bedrock AM100 at some distance from the source (stay clear from walls and other flat surfaces)

- Start the RT60 module; select IN (Interrupted Noise) in the settings screen.
- Go to the main measurement screen and press "arm."
- To trigger the measurement, shut off the noise source.

Once the device has been armed, it will start measuring the ambient noise field. At this stage the device is continuously sampling the audio, but the RT60 measurement itself has not yet been started. The device is merely detecting the moment a noise source has been shut off (in case the interrupted noise method is being used) or the occurence of an impulsive sound (in case the impulse method is used).

⚠ Make sure to select the correct method in the settings screen of the module (IN for interrupted noise, IR for impulse-response based measurements (e.g. with a starter gun). The description of the module in this section of the manual will be based on the interrupted noise method. We will indicate the differences for the IR method at the end of the section
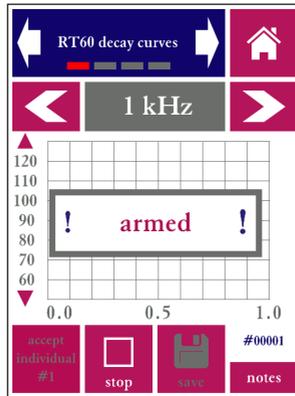
RT60 measurement screen, armed for measurement, but before a measurement has started

⚠ Make sure that the external sound source is always switched on BEFORE you arm the AM100 for a measurement by pressing "arm"

Once the beginning of the energy decay curve has been detected, the device starts measuring. By definition, no RT60 results can be calculated until the measurement and analysis are completely finished. While the device is acquiring measuring data, the following screen is displayed:
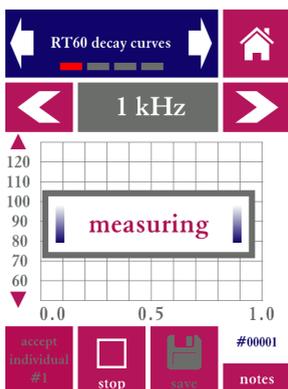
RT60 measurement screen, while the measurement is running and data is being acquired

Once the measurement is completed, the energy decay curves are plotted. Use the arrows to cycle through the frequency bands (note that you can choose whether to measure in 1/1 or 1/3 octave bands through the "settings" screen).
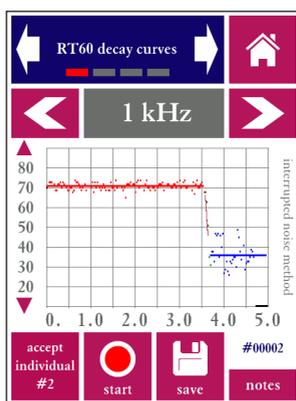
RT60 decay curves plotted once the measurement is completed (interrupted noise method)

Inspect the decay curves to visually verify that the measurement is correct. An incorrect measurement (which should be rejected) may become apparent in any of the following ways:

- The plotted data points do not reflect the expected linear decay of energy: the level does not decrease linearly over time. If this happens, the triggering process may have been disrupted.
- The linear decay curve that is fitted (drawn line) does not match the displayed data points. E.g., the starting point of the decay curve may be off.

- The signal-to-noise ratio (difference between the steady-state noise signal and the system noise floor) is insufficient.

From the decay curves, three different reverberation time metrics are calculated:

- The EDT (Early Decay Time), based on the first 10 dB of decay
- The T20, based on the section of the decay curve between -5 dB and -25 dB
- The T30, based on the section of the decay curve between -5 dB and -35 dB

Each of these three metrics has its own linear decay curve fit through the data points. The values of the EDT, T20 and T30 can be seen on the second screen of the RT60 module. Note that the reverberation time usually differs significantly between different frequency bands. Also note that (although respectively measured over 10, 20 and 30 dB of decay) the values of EDT, T20 and T30 are always extrapolated to 60 dB of decay; hence the name T60: the time it takes for the energy to decay to a level 60 dB below the starting level.

Cycle through EDT, T20 and T30 using the button on the right. Note that each of these three metrics is always calculated and saved for each measurement, even if only one metric at a time is displayed on the screen at a time.
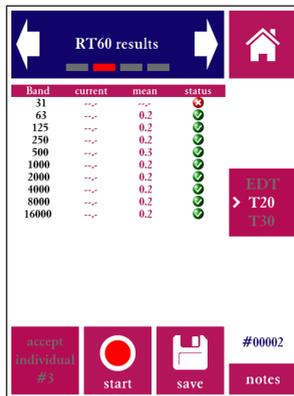
RT60 measurement results.

Quite often, a single RT60 measurement is insufficient to obtain a stable result. Many application standards require that data from multiple measurements are averaged. Sometimes the position of the noise source and/or measurement microphone must be changed in between measurements.

The averaging of multiple measurements is supported in the following way:
- When browsing through save measurements (in the "browse measurements" screen), you will see the mean data from each measurement series.

- On the bottom left of the screen you will find an "accept individual" button, which is not usually present in other measurement modules. Use this to commit a measurement to memory, if you are satisfied (upon inspection of the decay curves) that the measurement is valid.
- Once your first (valid) measurement is finished, press "accept individual"
- The letters and the numeral "#1" are now greyed out on the accept button.
- Now repeat the test procedure as many times as required, pressing "accept" after each measurement. The serial number is incremented after each individual measurement.
- Data from up to 20 individual measurements can be kept in the device memory at the same time.
- In the "RT60 measurement details" screen, you can inspect the latest individual measurement as well as the mean across all measurements.
- Once you are satisfied with the overall (mean) results, press the "Save" button (to the right of the "arm" button). Mean data from all measurements are now permanently saved and the temporary memory is reset.

The capability of averaging measurement data adds complexity to the user interface, but averaging is sometimes essential to arrive at valid end results. Note that averaging takes place on the level of the individualy measured data points along the decay curve (so not just across individually measured reverberation times).

In the table on the measurement details screen, a column is shown that indicates the status (reliability) of the current average measurement for the corresponding frequency:

- ✅ A green checkmark indicates a healthy, reliable measurement. The RT60 measurement data may be presumed accurate with a margin of error of at most 0.1 seconds.
- ⚠️ A yellow exclamation point indicates that the measurement accuracy cannot be guaranteed. This may have various causes. For example, the signal-to-noise ratio may be insufficient (not enough dynamic range between the steady-state noise signal and the noise floor with the noise source switched off). This may also happen if the signal fluctuates more than expected for a given frequency band and the resulting linear decay fit is poor.
- ❌ A red cross indicates that the result is statistically deemed unreliable. In order to obtain a reliable result in the corresponding frequency band, averaging across more individual measurements (adding to the overall measurement time) is needed. If the red cross persists upon averaging across multiple measurements, disregard the corresponding frequency band or repeat the entire measurement series.

It is good practice (and a formal requirement for many applications) to average across as many measurements as needed to obtain a green cross next to all relevant frequency bands.

⚠️ Never rely on results from RT60 measurements until you have checked the reported status per individual frequency band. Note that in some rooms it may be difficult, if not impossible, to obtain reliable results for the lowest frequency bands.

Please take note of the following practical tips and pointers for RT60 measurements:

- Do not place the measurement microphone and noise source too close to each other; if possible, estimate the reverberation radius in the room, and make sure that the microphone is as far outside the speaker's reverberation radius as possible.
- When using the interrupted noise method, make sure that your noise source is capable of producing levels far above the ambient noise floor in the room, and above the electronic noise floor of the device. Ambient levels of 90 dB(A) and higher are often needed. The SNR must be at least 50 dB in all relevant frequency bands. A spherical loudspeaker configuration playing pink or white noise is the preferred source of measurement noise. If you are using a single, non-spherical loudspeaker, avoid placement of the microphone on the loudspeaker axis. Even though the microphone is omnidirectional, avoid pointing the microphone straight at the loudspeaker.
- When using the impulse response method, use a starter gun or "clapper" that is capable of producing high level impulsive sounds.
- Eliminate sources of background noise.
- The state of all doors, windows and other separations should be fixed (open or closed) throughout the measurements. The opening of doors and windows changes the reverberation time.
- Do not walk around when the measurement is running, especially not in the immediate surroundings of the noise source.
- The signal-to-noise ratio requirements for RT60 measurements are sometimes difficult to meet. Especially for T30 measurements, it may be necessary to play back the test stimulus (interrupted noise) at high sound pressure levels.
- If your signal-to-noise ratio is insufficient for reliably measuring the higher frequency bands, consider switching over to white noise (instead of the commonly used pink noise) as a source stimulus.

If you are having trouble getting acceptable measurement data (green balls with check marks next to your measurement data), it may help to study the exact reasons why your measurement data is marked for rejection.

Along with each measurement value, the reliabily status is display (acceptable, warning, error). The stored data for each measurement (downloadable via USB) contains more details on the exact motivation for the indicated reliability status. This is indicated with the following letter codes:

- d: no data was present (error)
- r: not enough range (warning)
- b: bandwidth-time product too low (error)
- n: snr too low (warning)
- s: start time before knee point of the decay curve (warning)
- c: correlation of the decay curve fit is low (warning or acceptable, depending on value)

Multiple letter codes may be stored with a single measurement value.

The description in this section was based on the interrupted noise method. The procedure is similar for the IR-based method: after pressing "arm," generate the impulsive sound (fire a starter gun) to automatically trigger the measurement.

Note that the method used is indicated alongside the decay graph (in gray vertical text).
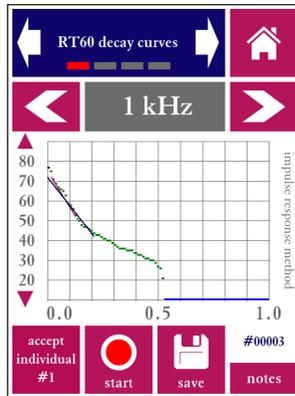
RT60 decay curves plotted once the measurement is completed (impulse response method)

9.2. Noise curves

The "noise curves" module implements a series of measurement methods to evaluate noise levels, based on sets of standardized spectral curves. The objective of these methods is to set criteria in terms of a single-number rating, while taking the spectral characteristics of noise into account.

The following methods are implement

Noise Rating (NR). This is an ISO-standardized (ISO-1996) method. Although the method is being pushed into the background by successive updates to the relevant standards, it is still seeing a fair amount of use, in particular in Europe.

Noise Criteria (NC). This method is frequently used in the United States, mostly for evaluating background sound in buildings and for specification of desired levels in advance of construction. Different versions of the NC have been in circulation since the 1950s; the AM100 implements the version standardized through ANSI S12.2:2008.

Preferred Noise Criterion (PNC). This is a variant of NC, not included in ANSI S12.2, but proposed originally by Beranek et al (1973) to yield a better correspondence with perceived annoyance by putting more emphasis on the lowest frequency bands and highest frequency bands.

Balanced Noise Criterion (NCB). Standardized through ANSI S12.2:1995, similar in purpose and application to NC; apparently intended upon its introduction to replace NC.

Room Criteria (RC). This method is primarily used to quantify noise from HVAC systems in buildings. In addition to the single number rating (e.g. RC-35) a spectral classification of the signal is also given: Neutral (N), Hiss (H), Rumble (R) and Rattle and Vibration (RV).

Room Criteria mark II (RCII). Updated version apparently intended to eventually replace RC curves.

The Room Noise Criterion (RNC) method is currently not implemented on the AM100.

Each method has its own series of preferred limits, depending on the purpose of the room in which it is measured. ANSI S12.2 contains table of recommend maximum NC and RC values, and additional information is available through an abundance of online resources. A full list of these recommend limits is beyond the scope of this manual.

As part of the calculations, various intermediate metrics are calculated including the Speech Interference Level (SIL).

Each method uses its own procedure for rating noise spectra based on the series of curves associated with the method. The so-called "tangency method" plays a central role for NC and related measures; the rating is effectively determined by the highest NC curve "touched" by the spectrum. The frequency band that touches the highest curve (the most dominant noise band) is indicated as part of the rating, e.g: NC-53(125Hz).

A full description of the rating procedures is beyond the purpose of this manual. Please consult the corresponding standards for this information.

Carrying out a Noise Curves measurement is similar to an SPL or RTA measurement: set the duration of the measurement and other parameters, and start the measurement.
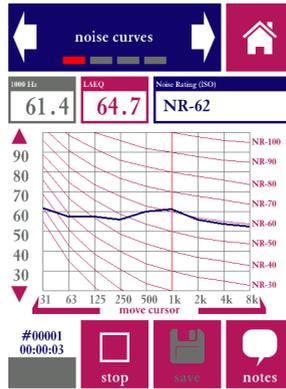
Noise curves. In this example, the curves and rating for NR are displayed

During a measurement, the reference noise curves are shown as red lines. The actual measured spectrum is shown as a blue line. The cursor can be moved around to read the SPL at any of the frequencies. The rating is shown, as well as the LAeq level.

Only one Noise Curves metric can be displayed at a time, but all metrics are calculated for every measurement. If a measurement is saved, the saved file contains data for all noise curve metrics. If you wish to evaluated more than one noise curve metric without measuring more than once, this can be done as follows:

- Select the first type of curves you wish to evaluate (e.g. NC)
- Carry out the measurement.
- Save the results
- Now change to another type of curves (e.g. NCB)
- Go to the measurement browser screen. NCB data will now be displayed
- Repeat this procedure to inspect results for further metrics (RC, RCII, etc).

The second screen of the module is used to set the duration of the measurement. Selecting curves (and the associated rating method) is done in the third screen of the module.
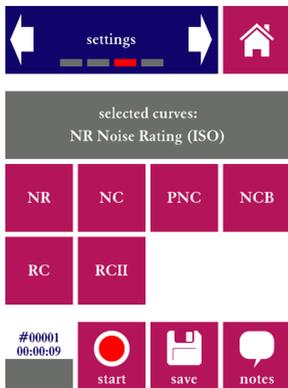
Choose from the available noise curve metrics

10. Electro-acoustics

10.1. Oscilloscope

The oscilloscope module is intended to quickly inspect acoustic signals, in particular periodic signals, in the time domain.
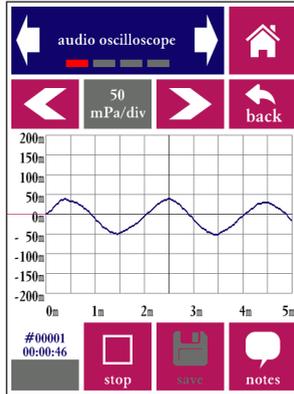
Sine wave with a frequency of 1000Hz as seen through the oscilloscope module

Directly from the first screen of the module (see above), you can control the horizontal scale (time base) and vertical scale of the graph, as well as the trigger mode. Select which setting you want to adjust, and press "set." Then choose the intended setting; by pressing "back" you can go back to reach the other settings again. Under trigger mode, you can set the level and slope (rising/falling). Red lines by the vertical axis indicate the trigger level.

Additional settings can be accessed through separate screens of the oscilloscope module. These include the measuring time, the choice whether or not to apply linear interpolation on the data, enabling or disabling frame averaging, and a choice between acoustic and electric measurements. If you are measuring sounds (with the microphone mounted), choose "acoustic." The amplitude scale will be expressed in the unit Pascal (Pa). If you are measuring without microphone, but directly into the XLR input, select "electric." The vertical scale will be expressed in Volts (V). Similar to any other oscilloscope, the settings for the time based and sensivity are expressed relative to the division lines on the screen (s/div, V/div, Pa/div).

⚠ Remember to turn of the 48V phantom power through the settings menu if you intend to do electric measurements. Note that the input impedance of the XLR input is (nominally) 2 kOhms.

The basic operation of the oscilloscope module is similar to the common (hardware) oscilloscopes that many engineers are familiar with. Note that the frequency response of the system is limited to audio frequencies (up to 20 kHz).

10.2. Polarity polarity

The polarity module is intended to check the polarity of loudspeakers. Polarity is important in all systems featuring more than one loudspeakers; inconsistent wiring may lead polarity inversions, with adverse consequences for the level and quality of sound. Note that inversion of the polarity by itself is usually not a concern, as long as the polarity remains consistent among all loudspeakers within the same space.
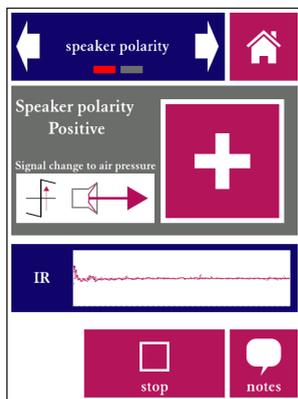
Loudspeaker polarity

Polarity measurements make use of test signals, usually saw-tooth wave patterns that either rapidly rise or decline in level. A test signal is provided on the flashdrive bundled with the AM100 and also available for download from the Bedrock audio website. The same test signal is featured on the BTB65 TalkBox (depending on the firmware version and production date of the talkbox). This default test signal is a sawtooth with a slow incline and a rapid descent and a period of 20 Hz. The module can also be configured for custom test signals, with a user defined period and up/down characteristic.

The polarity module is a configuration aid only; it does not have options to save data as most measurement modules do. The module is operated as follows:

- play the test signal continuously over the sound system under test
- start the measurement on the AM100
- bring the measuring microphone up close to the loudspeaker (typically 0.5 to 1m, depending on the acoustics of the room and the configuration of the loudspeaker). A little experimentation may be needed to determine the optimal measuring distance.
- The module shows a + or - sign to indicate the signal polarity, alongside a graphical representation of the loudspeaker cone movement in response to the test signal
- A graphical representation of the impulse response is also shown.

It can be difficult to obtain consistent readings for multi-way loudspeakers; it may be necessary to measure op close to each of the drivers.

10.3. Electric in RTA and FFT

Dedicated versions of the real-time analyzer (RTA) and FFT analyzer are provided specifically for analysis of electric signals through the XLR input.

⚠️ The precautions mentioned above (in section 9.2) also apply here: be careful to protect the XLR input from voltages higher than 40V, and make sure to switch of the phantom power in the settings menu.

The RTA and FFT intended for electric input signals are used in exactly the same way as the "ordinary" RTA and FFT (see sections 7.3 and 7.4), with one difference: the displayed levels are relative to an electric reference signal instead of the acoustic reference of 2.10^{-5} Pa.

The electric reference can be set to display all levels in dBU, dBV or dBuV (corresponding to reference levels of , respectively, 0.7746 V, 1.0000 V and $1.000*10^{-6}$ V).

Note that saved results are stored in separate folders (named eRTA and eFFT), separately from results for acoustic measurement data. The AM100 is calibrated for electric inputs to the XLR, completely separately from its acoustic calibrations. The electric calibration factor can be adjusted in the "Calibration" module.

11. Electric analysis

11.1. AC Volt meter

The AC Volt meter module is a simple tool to quickly assess the output level of line-level audio equipment. The line output of a device is connected directly to the XLR input of the AM100 (after removing the microphone). The AM100 then measures the signal level and expresses it in the following three units:

- V rms
- dBU
- dBV
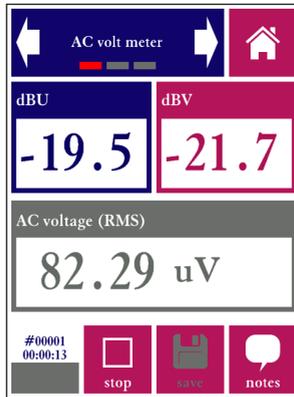
AC Volt meter

The XLR connector supplies 48V phantom power to the microphone. This phantom voltage can be switched on and off through the hardware settings menu. IMPORTANT: connecting equipment not designed for 48V phantom power to the XLR connector (with phantom power switched on) may permanently damage the connected device and/or the AM100. Disable phantom power before connecting to external equipment.

The "AC Volt meter" function of the device is intended for measuring audio output levels of amplifiers and other audio equipment. NEVER connect the XLR input pins to AC voltages over 40V; do NOT try to measure the AC mains with your AM100.

11.2. Fundamental frequency estimation (Hz)

The purpose of this module is to measure the fundamental frequency of periodic signals. In case of pure sine waves, the fundamental frequency is easy to measure and can be observed even with an oscilloscope. In case of signals with complex harmonic structures, estimating the fundamental frequency (which often corresponds to the perceived pitch) can be more challenging. This module is especially useful in cases where tones (or tone complexes) need to be tested against frequency or pitch specifications. It can also be used to tune musical instruments.
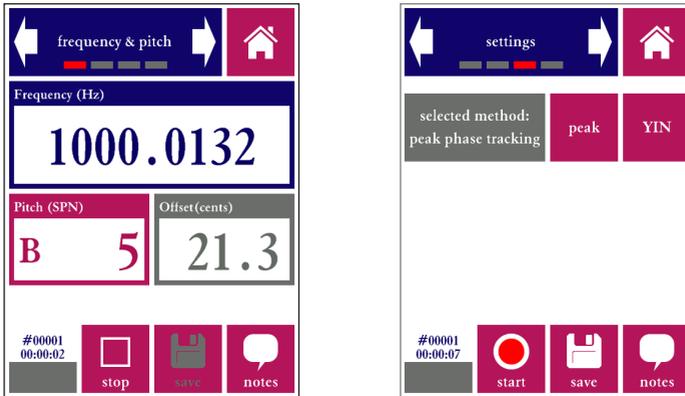
Measuring screen (left) and settings (right)

Two different estimation algorithms are supported: peak-tracking (peak) and YIN. Peak is more accurate in case of "clean" signals, while YIN is more robust against irregularities, and better equipped to deal with missing fundamentals.

The result of each measurement is the estimated frequency in Hz, as well as the corresponding pitch and the offset (in cents) between measured frequency and the closest tone according to the MIDI definition.

11.3. Total Harmonic Distortion and Noise (THD+N) THD+N

Total Harmonic Distortion and Noise (THD+N) is a metric intended to quantify distortion levels in electro-acoustic equipment. The value THD+N, which is expressed as a percentage (%) or a signal-to-noise ratio (dB) indicates the degree to which a system exposed to a single sine wave produces distortion and noise components, relative to the signal strength. Our implementation of THD+N follows the requirements of the AES17 standards. We use fixed-frequency sine waves (as opposed to sweeps or chirps) as test stimuli.

Measuring screen before setting a reference level (left) and after setting the reference level in dBU to 2.7 dB (right)

The procedure for performing a THD+N measurement is as follows:

- Present an undistorted sine wave to the input of the channel under test (e.g. using the BTB65 talkbox as a test source.
- Connect the AM100 analyzer to the output of the channel under test and press "start."
- The THD+N is now displayed as a signal-to-noise ratio (level) as well as a percentage. These two metrics are computationally equivalent.
- The output signal level from the channel (in dBU) as measured by the AM100 is also displayed; note that THD+N is highly dependent on signal levels.
- A reference level in dBU can be set. This helps to determine the maximum input level according to AES17 (by inspecting how changes in input level correspond to changes in output level).

Note that the test frequency assumed by the analyser must correspond to the sine waves at the channel input for the measurement to be accurate.

Through the settings menu, three options for settings the input signal frequency can be chosen:

- Automatic. The fundamental frequency is estimated from the output signal
- Snapped to 1/1 or 1/3 center frequencies. The fundamental frequency is estimated automatically, but rounded to the nearest 1/1 or 1/3 octave
- Fixed. The frequency is entered through a keyboard.
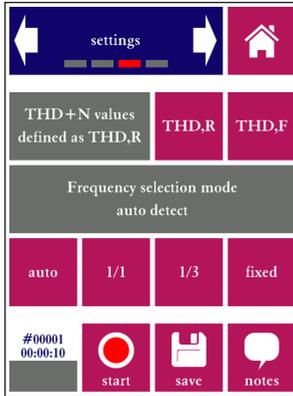
Settings screen of the THD+N module

Setting the test signal frequency is done through the settings menu, which also lets you choose between two different definitions of THD+N: based on THD_R (default) or THD_F. In case of THD_R, the total signal level is used as a reference for computing THD+N. In case of THD_F, the reference is the estimated level of the fundamental frequency only. AES17 calls for THD_R, while THD_F is perhaps a more intuitive definition that is also used by various other measuring instruments.

THD+N constitutes a relatively complex subject matter, a full description of which is beyond the scope of this manual. We recommend studying online reference materials and tutorials on THD for more information.

12. Building acoustics

The Bedrock AM100 features built-in functions to evaluate the airborne sound insulation as well as impact noise transmission. The modules on the AM100 aim to support measurement that comply both to ISO and ASTM standards. Four modules are provided: for impact noise and airborne noise, and for each respectively frequency-dependent values as well as single-number metrics.

While many manufacturers rely on separate software packages for building acoustics (running on a PC), the Bedrock AM100 modules require only the instrument itself to analyze sound insulation. For each room/space, multiple measurements are required, from which an overall transmission metric is computed. This makes the sound insulation modules different from other modules in the instrument, which are geared more towards single measurements. The advantage over the use of PC software is that results are immediately available while measuring in the field.

We recommend getting acquainted with the building acoustics modules through a series of test measurements before using it in a real-life test setting.

Two modules are provided for airborne sound insulation:
- **DnT/R':** frequency dependent airborne sound insulation between rooms according to ISO-16283-1:2014 (Dn, DnT and R') and ASTM E336.
- **SRI / STC**: single-number rating of airborne sound insulation between rooms according to ISO-717 (SRI) and ASTM E413 (STC).

Two modules are provided for measurement of impact noise:
- **L'nT:** frequency dependent impact noise measurement (while using a tapping machine) cf ISO-16283-2:2015.
- **L'nw/L'nTw**: single-number rating of impact noise according to to ISO-717-2:2021.

12.1. Airborne insulation: DnT/R'

NOTE: to evaluate the transmission loss (TL) as defined by ASTM E336 and ASTM E90, please adopt D values rounded to 1 dB. For simplicity, given that the standards are otherwise functionally identical, we will adhere to ISO definitions only.

The general principle for these measurements is that sound is generated in one room (the sending room) and then measured on either side of the partition (in the sending room and one or more receiving rooms). The difference between both (frequency-dependent) levels quantifies the insulation offered by the partition.

Four different frequency-dependent measures of sound reduction are assessed:

- **D: Level difference.** The difference in sound level (per frequency band) as observed between sending and receiving room (also known as Transmission Loss).
- **Dn: Normalized level difference.** The difference in sound level (per frequency band) is normalized to the size of the room. Note: *Dn is not specified in ISO-16283-1 but used in other standards.*
- **DnT: Standardized level difference.** Level difference D is standardized to the reverberation time of the receiving room.
- **R': Apparent sound reduction index.** Quantifies sound power observed in the receiving room, including direct as well as flanking incidence and other components. Takes the reverberation in the receiving room and the surface area of the partition between sending and receiving room into account.

The apparent sound reduction index R' can be compared to values of the sound reduction index R often specified in manufacturer's documentation for partition materials (obtained through lab tests). Under specific and controlled lab conditions, this module can also be used to measure R (instead of R'). Note that flanking transmission will lead to differences between the observed R and R'.

Please refer to ISO-16283-1 and/or online resources for further guidance on the procedures to be observed for obtaining reliable and repeatable test results. A full description of the procedure surrounding sound reduction measurements are beyond the scope of this manual. Please also refer to the standards for the formulae used to compute each of the metrics.

From a practical perspective, the procedure to determine D/Dn/DnT/R' for a room is as follows:
- Place a sound source in the sending room. Consider that the sound power in all relevant 1/3 octave bands (at least 100 Hz - 3150 Hz, preferably 50 Hz - 5 kHz) has to be high enough to overcome the sound insulation of the partition and still be above the lower detection threshold of the analyzer in the receiving room.
- Measure the 1/3 octave spectrum at one or more locations in the sending room
- Measure the 1/3 octave spectrum at one or more locations in the receiving room
- Determine the mean D across sending/receiving measurement
- Measure the 1/3 octave spectrum of the natural background noise in the receiving room; reject if too loud (or apply corrections per the standard).
- Calculate Dn and DnT by normalizing/standardizing in terms of the reverberation characteristics of the receiving room.
- Calculate R' based on D taking into account the surface area of the partition as well as the reverberation time in the receiving room.

Each insulation measurement reported by this module is indirectly calculated by a series of RTA measurements (with 1/3 octave resolution). The basis for each calculation is the difference, per 1/3 octave band, in the level between sending and receiving room (D) followed by normalizations (Dn, DnT, R').

When the module is started, a table of RTA measurements related to one partition is displayed, as shown below.
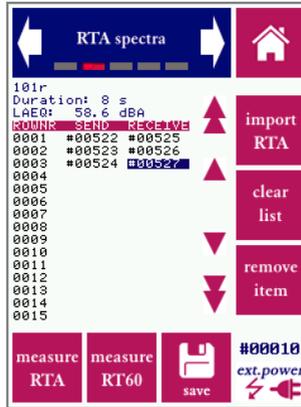
Table of RTA measurements from which the insulation results are calculated.

The table will be empty when the module is started for the first time. Entries in the table will be retained until the list is cleared (by pressing "clear list") even if the analyzer is restarted. The left column contains measurements for the sending room, the right column for the receiving room. To obtain reliable results, averages need to be taken across multiple RTA measurements for the sending as well as receiving room (per most standards, the minimum number of measurements in both rooms is 5).

The sending or receiving column can be selected by pressing the corresponding column. Entries can be added to the table by pressing a position in the table and then pressing "Import RTA." The measurement can then be selected from all RTA measurements stored on the instrument (previously measured). A new RTA measurement can be started by pressing "Start RTA measurement."

⚠ For sound insulation measurements, the applicable standards mandate that measurements are obtained in 1/3 octaves. RTA measurements obtained with other bandwidths (1/1, 1/6 and 1/12) cannot be used in this module.

Note that measurements used for the sending room cannot be used for the receiving room, and vice versa (as this would result in incorrect results). The module does not allow the same RTA spectrum to be used for sending as well as receiving room.

⚠ If RTA measurements that were obtained with different sound source characteristics (or placements) in the sending room, make sure that the number of measurements per loudspeaker configuration is equal between sending and receiving room (e.g. 5 sending and receiving RTA measurements for each configuration).

Based on the table of RTA measurements, the module calculates the value of D for each frequency band. However, for the other metrics to be calculated, the reverberation time must also be known,as well as the background noise spectrum in the receiving room. The 3rd screen of the app is used to select a stored RT60 and RTA measurements; use the keyboard to input the measurement number of a measurement for the receiving room. Alternatively, a new measurement can be started (by pressing the button "measure RT60" or "measure RTA" to start a background noise measurement). The reference RT60 as defined in ISO-16283 (by default 0.5s) can also be set.

⚠ Only reverberation time measurements obtained with 1/3 octave band resolution can be used by this module.

In addition to RTA measurements and the reverberation time, additional information about the receiving room and the partition between the rooms is needed to calculated DnT and R'. The fourth screen in the module allows the user to enter this information.
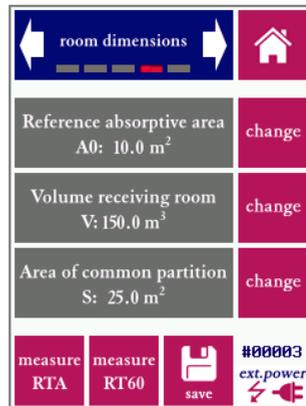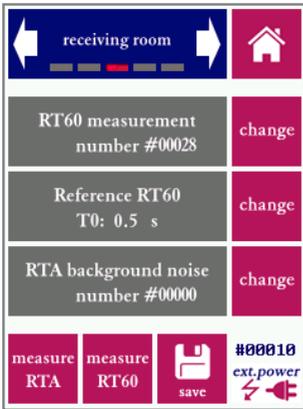
Screen 3 (receiving room RT60 data) and screen 4 (room dimensions)

As with other AM100 modules, screen 1 is where the results are shown. Unlike other modules, the module when first started from the menu shows screen 2 instead of screen 1 (since RTA measurements need to be chosen before any results can be displayed).

If no reverberation time measurement has been selected, only a curve is shown for D on screen 1 (since this measure does not require reverberation times in the receiving room to be known). If an RT60 measurement was selected, four curves will be shown in four different colors, corresponding to D, Dn, DnT and R'.

As always, the "save" button is used to store the measurement data as displayed on screen 1. Note that the table of RTA measurements is not automatically cleared after saving results, so that it is possible to modify the table by adding/removing measurements and saving the outcome as a new insulation measurement result.
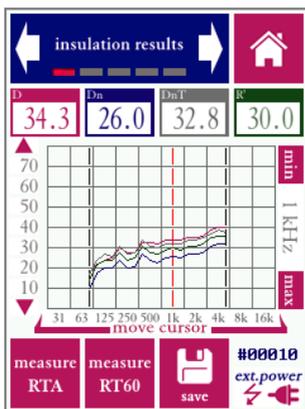
Results: sound insulation measures computed from RTA and RT60 measurements.

New RTA and RT60 measurements can be started from any screen. Once the measurement is completed, the "back" button on the top right of the RTA and RT60 module (where otherwise the home button would be shown) takes the user back to the Insulation module. However, note that any new measurements will not be included in the calculation until they are added to the RTA table or selected as the RT60 measurement.

The D, Dn, DnT and R' value for any specific 1/3 octave band will be displayed at the top of the screen after moving the cursor to the corresponding band. The buttons "min" and "max", shown at the right side of the screen are used to set the maximum and minimum frequency for which the measured metrics are displayed (typically 100-5000 Hz). To set the measurement range: move the cursor to a frequency band and subsequently press "min" to set the minimum or "max" to set the maximum. Calculated results can be saved, and previous results can be browsed, in the same fashion as all other modules.

⚠ In some cases, the curves are not shown for all frequency bands. In those cases, usually some lower and/or higher bands are the missing. There are two reasons for this to happen: either the RT60 results for the missing bands are flagged as unreliable, or the background noise level in these bands ais too high. Corrections for elevated background noise levels per ISO-16283 are applied automatically, but if the background noise is les than 6 dB below the signal the measurement must be rejected altogether, and no results can be displayed.

⚠ Results are saved in a common file format for the (frequency-depedent) insulation module (D, R') as well as the single-metric SRI/STC module (in the folder BUILDING). The information that is displayed on the AM100 is different for both modules, but both single-metric as spectral results are stored with every measurement/calculation done on the device.

12.2. Airborne sound insulation: SRI / STC

Based on the same (frequency-dependent) measures used in the sound insulation module, single-number metrics can be derived. ISO-717 defines the Sound Reduction Index (SRI), indicated as R'w. ASTM E413 defines the Sound Transmission Class (STC), which is conceptually similar both computed somewhat differently.

The single-number metrics R'w (SRI) and STC are used most commonly when specifying and measuring the sound insulation offered by partitions within buildings. In both cases, the approach is to define a specific reference curve that is shifted until it matches the measured curve. The criterium for matching differs between SRI and STC, and the observed frequency range is also slightly different. Four different single-number metrics can be measured by this module (use the arrows to select the current metric):

- **R'w: Weighted apparent sound reduction index.** This is the dominant metric used when applying ISO-717 to measure and specify sound insulation in buildings.
- **Dn,w: Weighted normalized level difference.** The difference in sound level (per frequency band) is normalized to the size of the room (ISO-717).
- **DnT,w : Weighted standardized level difference.** Level difference D is standardized to the reverberation time of the receiving room (ISO-717).
- **STC: Sound Transmission Class.** A weighted, single-number measure of sound attenuation as defined by ASTM E413.

In screen 1 of the module, the level difference D (called Transmission Loss TL in ASTM E413) is shown in a graph, together with the level-shifted reference curve that is shifted to match the level difference.
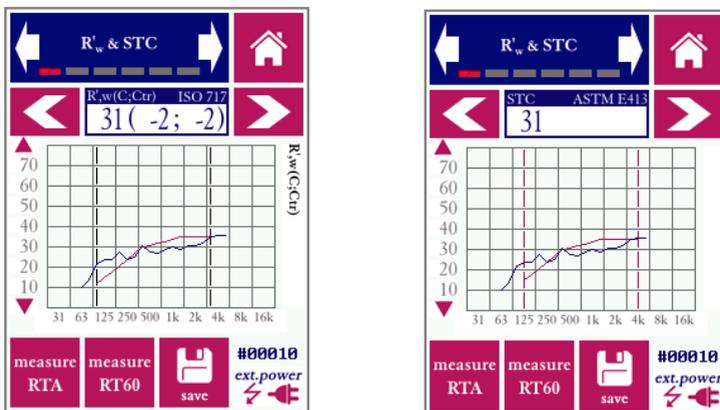
R'w (left) and STC (right) measurement result based on the measured D and the shifted reference curve (show in graph). Note that first RTA spectra need to be selected in screen 2 before these data can be calculated.

The procedure for measuring and selecting RTA spectra and setting receiving room characteristics is identical to the "insulation" module (refer to section 11.1 of this module for details).

⚠ The four different single-number metrics cannot be displayed at the same time, but they are always calculated and stored in the saved output file. To review a different measure (for the same D-values), simply save the result, change the settings, and then revisit the saved file in the "browse measurements" screen. Note that SRI / STC values can also be observed by reviewing data previously saved in the "Insulation" module.

ISO-717 does not just define the primary metrics mentioned above (R'w, Dn,w and DnT,w) but also two so-called "spectrum adaptation terms," C and Ctr. These terms are used in some applications and countries to correct the primary metrics. The results of ISO-717 measurements are displayed in the format defined in the standard, e.g.

$$R'w \ (C \ ; \ Ctr)$$

for R'w (and similarly for Dn,w and DnT,w).

Application standards call for different frequency ranges to be used for the spectral adaptation terms. These ranges can be set in screen 5 of the module. The graph displayed on screen has a caption on the right side (printed sideways) that shows the frequency limits applied for the spectral adaptation terms.

⚠ If no values for C and Ctr are shown, this means that no reliable RT60 data was available for all frequency bands. Obtaining RT60 readings down to 50 Hz is a challenge in many cases; if data is not available for all bands, no values for C and Ctr are displayed.

Note that R'w, STC and the other supported metrics can be quickly evaluated in the field using nothing but the Bedrock AM100: no supporting software is needed. For reporting (and deeper analysis), it may still be beneficial to download the saved data on a PC and further investigate the data in Microsoft Excel. Note that we provide MS Excel tools for reporting and analysis purposed for almost every module on the AM100. These Excel tools can be freely customized to meet your own specific needs and purposes.

12.3. Impact sound insulation: L'nT

The propagation of impact sound in buildings is usually evaluated by means of a tapping machine. The procedures for this are described in standard ISO-16283-2:2015.

Similar to airborne sound, there are frequency-dependent metrics (Li, L'n and L'nT) and single-number metrics, which we support in a separate module (see section 12.4).

The metrics measured through this module are defined as follows:

- **Li: Impact sound pressure level within a 1/3 octave band.** This is the directly measured frequency-specific level measured due to the tapping machine in the sending room, as measured in a receiving room.
- **L'n: Normalized impact sound pressure level.** The level in the selected 1/3 octave band, normalized to the size of the room.
- **LnT : Standardized impact sound pressure level.** The level in the selected 1/3 octave band, standardized to the reverberation time in the receiving room.

Note that airbone sound insulation measurements result in transmission loss values (effectively a difference in level due to the building structure). Impact noise measurements assume a standardized tapping source in a sending room, and measure the resulting sound level in a receiving room. Otherwise the procedures are similar. Multiple measurements in the receiving room are usually included to arrive at combined (averaged) results. Consult the ISO standard for the detailed procedure - there are several variants. Note that the so-called "rubber ball method" is currently not supported.

The basis for the calculations is a list of RTA measurements.
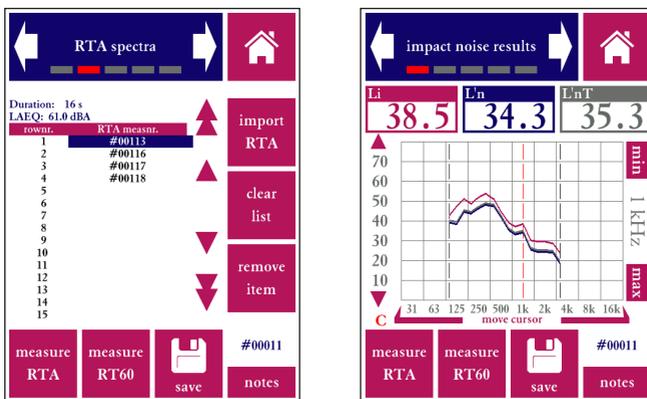
List of RTA measurements for the calculation (left) and results from L'nT measurements (right). By moving the cursor, individual values for each 1/3 octave band can be read.

12.3. Impact sound insulation: L'nw / L'nTw

The single-number impact noise metrics according to ISO-717-2:2021 are implemented in this module. The same data structure is used as for the L'nT module, but a single number metric is calculated in this case from the spectral data. Alongside L'nTw the corresponding adaptation values are calculated. Please refer to ISO-717-2 for measurement instructions and for guidance on interpration of the data.

Settings are provided to adjust the frequency range for which adaptation values are calculated (starting at either 50 Hz or 100 Hz). In some cases, the calculated adaptation values can potentially be negative. An option is given to clip negative values (and set them to 0 in the displayed results.
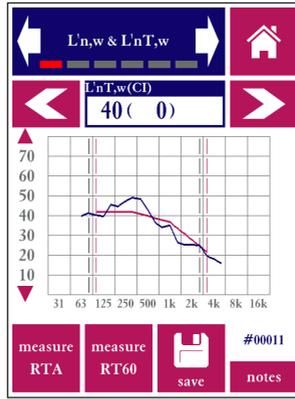
Results from L'nTw measurements. LnW or L'nw measurements can be shown alternatively

Similar to other building acoustics modules, the characteristics of the room (reverberation time, volume, etc) must also be known. RT60 and background noise can be measured; a measurement number is entered as a reference for the calculations. The room volume must be entered manually.

Note that ISO-16283-2 and ISO-717-2 provide specific instructions on the number and location of RTA measurements. This manual only provides information on how to use your Bedrock AM100 to perform the described measurements. We recommend to use the original standards as a reference when planning and performing measurement.

The overal procedures for measuring impact noise are analogous to airborne insulation measurements (except that a tapping machine is used instead of an airborne noise source, and that no source RTA measurements in the sending room are needed). If the procdure for impact noise is unclear, please refer to sections 12.1 and 12.2 for the airborne noise procedures first, since these sections explain the process in more detail.

13. Other modules and control modes

13.1. Calibration

The Bedrock AM100, the instrument itself as well as the microphone, is calibrated before leaving the factory. A calibration certificate is included with the device. However, it is recommended that you carry out your own "user calibrations" from time to time, especially under the following conditions:

- You are using the device considerably above or below room temperature
- You are planning to use the device again after a prolonged period of storage
- You are planning to do measurements for which a high degree of accuracy is required, e.g. certification measurements

Reference sources (calibrators) are available from a number of vendors. These generate a test tone which is adaptively adjusted to a fixed, accurately know signal level (usually 94.0 dB). This makes calibration relatively easy: insert the microphone tip into the port of the calibrator, using an adapter with suitable dimensions for the microphone tip. The device should now measure the nominal signal level produced by the calibrator. Any type Class 1 /Type 1 calibrator can be used. We recommend the Bedrock BAC1.

The Bedrock AM100 can be calibrated with any stable sound source of a known reference level. This can be a calibrator, but a other non-fluctuating sound field will also work.
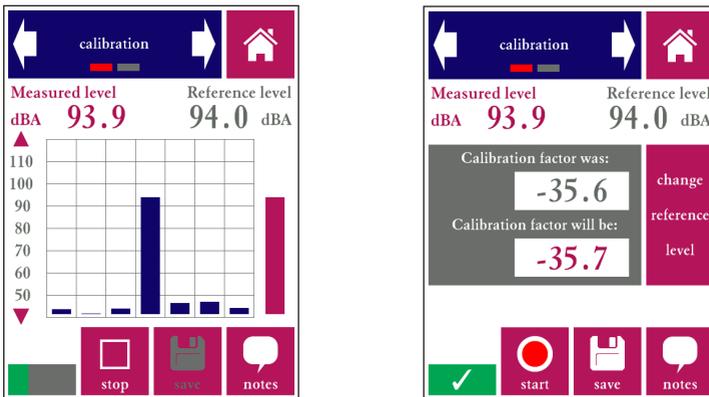
Calibration module while measuring a 1 kHz calibration tone (left) and once the calibration factor has been calculated at the end of the procedure (right).

When calibrating the AM100 with BAMT1 microphone, correct the nominal value of the calibrator (typically 94.0 dB) with -0.1 to account for the free field response and other correction factors.

To start calibrating, apply the calibrator to the microphone and press "start." Do not worry if the reference level shown is incorrect; you can change this after the measurement. An octave band spectrum will be shown while the equivalent-continuous A-weighted sound level is measured. Once this measurement is finished, the Bedrock AM100 will show the results.

The new calibration factor (sensitivity adjust) of the device is shown, as well as the original calibration factor as currently used in the device. The differences between the old and new value are typically small (less than 5 dB), unless you have connected a different signal source or microphone than the device is currently calibrated for.

If you already know the calibration factor for your microphone, you can enter it manually. The second screen of the calibration module has two editable fields: "acoustic calibration factor" and "electric calibration factor." The acoustic calibration factor is the same parameter that is normally measured with a calibrator.
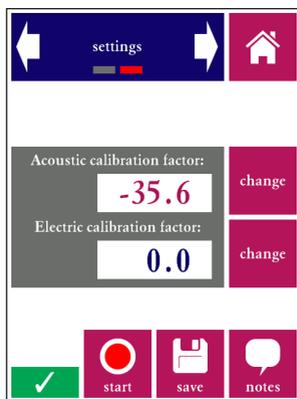
The second screen lets you enter calibration factors manually.

The electric calibration factor cannot be determined automatically with a calibrator; it can only be changed by hand. This parameter modifies the sensitivity of the XLR input for electric measurements (e.g. AC Volt measurements). Recalibration should rarely, if ever, be needed; exercise great caution when adjusting this value.

We recommend that you verify any new calibration in the SPL meter module. Re-apply your calibration signal source, and check that the correct reference level is measured.

⚠ Once you press "save," the measured calibration factor is committed permanently to the device's non-volatile memory. This overrides earlier calibrations. Do not manually change calibration factors unless you are absolutely certain that this is what you need to do. All measurements will be affected.

13.2. Settings settings

The settings module allows you to configure the overall settings of the device, in addition to the module-specific settings found in each module. The following attributes can be set:

- Time and date (time in 24h format and date in DD-MM-YYY format)
- Phantom power to the microphone
- Microphone type selection
- Microphone pre-amplifier gain (measuring range)
- Time interval before the backlight automatically switches off to save power
- Brightness of the display (*not available on early versions of the SM50*)
- Recording of audio during measurements (on / off)
- Text annotations with each measurement (on / off)

The system date and time are changed by means of the arrows (up/down). Note that your changes are not applied until you press "set clock."
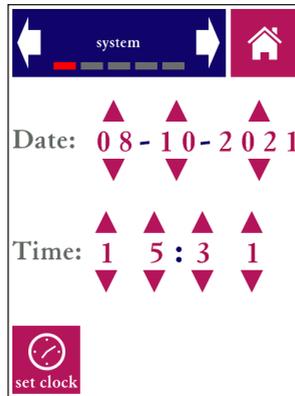
Date and time in the settings module

The Bedrock AM100 has an internal real-time clock that keeps running even when the device is powered off. The clock needs to be adjusted if the battery has been depleted completely, for instance when the device has not been used for a very long time. The system date and time are used for time-stamping the saved data. If you download measurement data over USB, the file creation date/time are only accurate if the internal real-time clock of the device was updated to the right date and time.

From the "microphone" screen, the 48V phantom power can be switched on and off. If the AM100 is used with its own (bundled) microphone, phantom power should be switched on. Turn the phantom power off when measuring (line) levels from external electronic devices.

⚠ IMPORTANT: connecting equipment not designed for 48V phantom power to the AM100 XLR connector (with phantom power switched on) will permanently damage the connected device. Use this setting with discretion. When in doubt, turn the phantom power off.
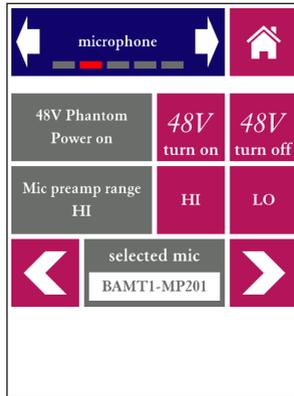
The microphone and preamp settings: turning phantom power on/off, selecting the preamp gain (measuring range), selecting the microphone type, and selecting free field or diffuse field measurements.

The internal microphone amplifier of the AM100 features an adjustable gain setting. Adjustment of the pre-amp gain is rarely needed, since the system has a wide (linear) dynamic range. However, if you wish to connect an external signal source which operates at much higher signal levels than the standard microphone, you can choose adjust the internal pre-amp gain, by choosing between two ranges: HI and LO (default). We recommend switching to the HI range when doing high-level (impulsive) sound measurements higher than 115 dB.

Different models of the AM100 may ship with different microphone/preamp types. Each microphone model has its specific underrange/overload limits in each of the two measuring ranges (LO and HI). By selecting the correct type of microphone here, the parameters of the device are set to the right values.

Always make sure that the microphone selected here matches the microphone mounted on the device. The option "XLR electric in" is used when no microphone is used, but the signal come from an external source into the XLR connector. This option sets the overload/underrange settings to the limits of the internal preamp electronics, and sets all free-field/diffuse field corrections to zero.

If you plan to use a third-party microphone with your AM100, you will need to obtain a so-called "mic.ini" file for that particular microphone. Please contact support if you need access to third-pary microphone data (if available).

The settings menu also features a screen that lets you configure the power saving settings of the device, and a screen with settings for delaying and repeating measurements.
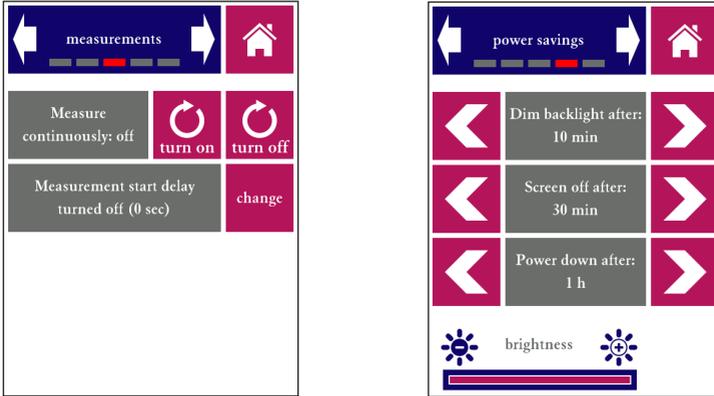
Left: measurement settings (measure continuously and start delay
Right: power saving settings (backlight time-out and display brightness)

When "measure continuously" is enabled, the AM100 will keep starting new measurements until manually interrupted. This applies throughout all modules. The same is true for the option "start delay," which can be set from 0 seconds (off) to 60 seconds. This option is useful to allow time for (for example) leaving the room or stepping away from the meter.

The backlight time-out makes the backlight automatically switch off if the touch screen has not been touched for a while. This time-out can be set between 30 seconds and 30 minutes, or the backlight can be set to never turn off at all.

The backlight time-out option saves battery power. But keep in mind that it will cause the screen to black out *without turning the device off.* Don't forget to turn the device off when you finish working; the fact that the screen is dark does not necessarily mean that the device is turned off. To turn the backlight back on again after a time-out, press the power button. The touch screen will not respond while the backlight is turned off, but measurements will continue to run.

The fifth screen of the settings module ("misc settings") contains two settings: audio recording and text annotations.
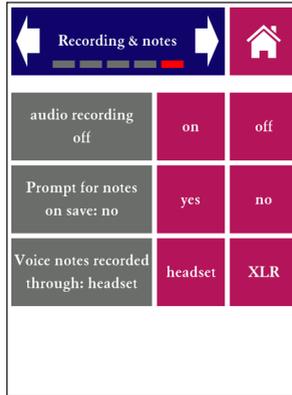
Audio recording and text annotations.

With audio recording set to ON, the audio of each and every measurement is recorded onto the internal storage of the AM100 in .wav format, calibrated to 150 dB SPL at full scale. Note that this uses up significant amounts of storage space. Once enabled from the settings tab, audio will be stored whenever saving a measurement from any module, until audio recording is turned off again from the settings menu. If audio is being recorded, a tape recorder symbol (⊙⊙)will be shown above the progress meter during measurements.

With "prompt for notes" set to "on", the text/voice annotations screen is automatically brought up whenever a file is saved. Of this is set to "off" then the user has to manually press the "notes" button first. Automatically prompting for notes prevents the user from forgetting to enter a note, but is less convenient if notes are only required once in a while.

Voice notes can be recorded either through the main microphone (in between measurements) or through the mic channel on the headset jack.

13.3. Network

The Network module is intended to connect the AM100 to a local area network via WiFi. WiFi can be enabled/disabled through the "settings" menu from the top bar in any module, but before WiFi can be used, network configuration settings need to be set.

The module has two screens. From screen one, you can select one of the available networks (SSID). You can scan for the available networks, select a network, or forget the selected network. Screen two shows information related to the AM100 on the wireless network, which you will need in order to connect to the AM100 from another device (computer) on the wireless network.

Network selection

EmbeddedAcoustics_DMZ_2.4

SSID	strength
EmbeddedAcoustics	-68
EmbeddedAcoustics_Guest	-68
ziggo-ap-5449181	-69
EmbeddedAcoustics_DMZ_2.4	-71
PHYSEE_FPF_2.4G	-81
H369A56BA5C	-84

scan   connect   forget

Network status

MAC address:	60:09:c3:0e:f5:f8
IP address:	192.168.178.97
Remote control port:	23
Remote control key:	z9MX3k
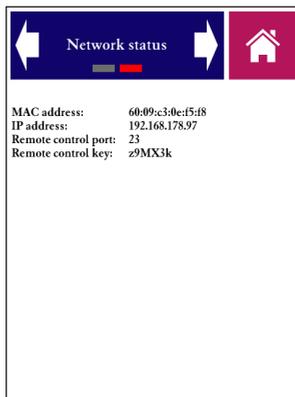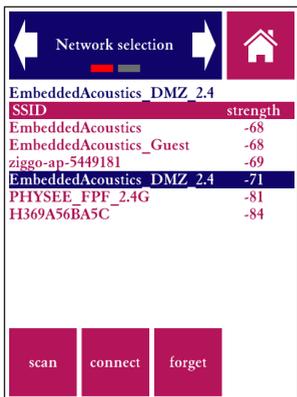
In order to connect via the network, you need your AM100's IP address and its remote acces port. Please refer to chapter 14 (and separate documentation for our PC software Bedrock Studio) for details. The remote control key is included as a security feature to prevent unauthorized applications on the network to control the AM100 without your permission.

13.4. Status

System status information can be read using the status module. If you contact Bedrock Audio for technical support, you will be asked to provide some of the information displayed here (such as the firmware version and hardware revision). The current capacity of the battery pack is also shown here, as well as the number of charge cycles to date, he remaining data storage capacity and several other statistics related to the battery pack.

The status module is not used to make changes to the system, but just to retrieve system information.

The following information is displayed:

Mainboard serial: unique ID of the device mainboard
Mainboard revision: hardware revision ID of mainboard
Software version: version of current firmware (all modules)
SPL version: version of SPL firmware (IEC-61672)
Software ID: identifier for the type of firmware needed
Disk space free: free storage in MB
Battery capacity: total capacity in mAh, taking battery wear into account
Charge cycles: number of times the battery has been charged
Incomplete cycles: number of interrupted charge cycles
Self-discharge: estimated self-discharge while not used
Charges since cal: number of charges since the battery gauge was last calibrated

13.5 USB audio device mode

When starting this module, a special mode of the device is activated, in which it performs no other tasks than simply streaming audio to a PC while applying the AM100's level calibration. This is done in the form of a single-channel stream, at a fixed bandwidth of 48 kHz and a bit depth of 32 bits. All current versions of MS Windows should recognize the AM100 while operating in this mode as a "USB audio device" and may also indicate that it is an "omnidirectional microphone." Once the device is detected, make sure that the software that you intend to use is also configured for the device. A small Windows test application is included on the USB flash drive that came bundled with your Bedrock AM100.
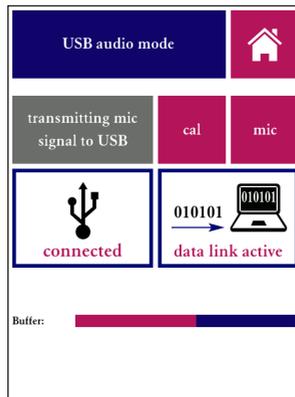
The SM50 display while in USB audio mode

Once the device is connected via USB (and the PC has recognized it as an audio device), an indication showing "USB connected" will appear. Once data starts streaming, an animated picture and the message "data link active" will indicate that data is being transferred. This will happen once an application initiates streaming audio from the Bedrock AM100.

The audio signal is scaled so that full scale corresponds to 150 dB (e.g, a signal with a level of 94 dB will correspond to a level of -56dB relative to full scale). Windows drivers and applications may digitally adjust the level by applying software-based volume settings. **If left unadjusted, the volume of the signal when directly played back on the PC may be quite low.** This is done on purpose, so that high levels of up to 150 dB can be accommodated. Given the fact that 32-bit audio has a dynamic range of 192 dB, this does not affect the integrity of even the lowest-level signals. The entire dynamic range of the Bedrock AM100 is available digitally. If the audio is too soft, simply adjust the volume control on the PC.

If you want to explicitly calibrate your windows software to work with the Bedrock AM100, press the "cal" button to generate a synthetic 1000Hz sine at 94.0 dB. If you calibrate your software to this level, then all audio measured by the AM100 will be registered at the correct levels in your software as well. Press "mic" to return to normal operation.

14. Accessing saved data

14.1. Connect USB cable

For the moment, saved measurement data, audio recordings and screenshots can only be downloaded through USB. We aim to add WiFi support in a future firmware upgrade.

When you connect the Bedrock AM100 to the USB port of a PC or Mac, a dialog window will appear on the device, asking if you wish to start data transfer. If the AM100 is powered off when you connect it through USB, it will power on automatically.

If your AM100 is already phyiscally connected (and powered) through USB but not showing up on your PC, open the top bard settings menu (gear icon on the top right) and press "connect."

The Bedrock AM100 will present itself to the USB host as a mass storage device. This allows you to access the internal file system of the AM100. You will see these directories/folders:

- UPDATES: place update packages here to update the firmware upon reboot
- CALIBR: Calibration data is saved here
- LOG: System logs are stored here (e.g. battery life cycle management)
- SCRNSHOT: Screenshots from any module are saved here
- SPL, STIPA, RTA, FFT, RT60, LAEQ, SCOPE, USBAUDIO, NOISE-CUR, ACV, SPEECHLM: folders where measurement data and recorded audio from each of the measurement modules is stored.

IMPORTANT: the AM100 system files are kept in a separate partition that is normally not mounted when you connect a PC. If you need direct access to your system folder, contact Bedrock support to unlock your system partition (which will show up as a separate drive letter). Use extreme caution when altering system files. *Please note that service necessary due to changes applied to files in the system partition is not covered by warranty.*

14.2. Copy your measurement data

In order to process and analyze your measurement data further, copy all results files from the Bedrock AM100 to a local folder on your PC. The measurement files are saved in .csv format. These files can be imported in almost all worksheet applications, such as Microsoft Excel. MS Excel templates for downloading and processing your measurement data are provided on the flashdrive included with your device. These will allow you to analyze the data further, and pre-process the data for the purpose of reporting. User manuals for these templates are embedded in the MS Excel files themselves.

14.3. Resetting measurement number series

Even if you completely remove the measurements from the device, the AM100 will continue the measurement numbering series where it left off. In order to prevent duplicate numbers, the counter is not automatically reset. If you wish to reset it manually (restarting at number 1), you need to remove all .csv files. as well as the file named _INDEX.DAT.

14.4. About the file format

Note that the .csv format is not a formal standard; differences in interpretation of the .csv file format exist even between different versions of MS Excel. This results in the fact that the output files from the AM100 may be imported very easily in some versions of Excel, and may require more attention in other versions of Excel. If you are having problems importing the output files into Excel, import the files as "text files" rather than .csv.

14.5. Downloading recorded audio

If the "record audio" option is turned on in the system settings, then all audio recordings are saved in the same folders as the .csv files with measurement data.

- Audio is stored in .wav format (single-channel 48 kHz floating point PCM), which is easily read by many applications
- The audio is normalized to exactly 150.0 dB at full scale.
- The level calibrations are implicitly preserved by this normalization relative to full scale. This allows for easy further processing by acoustic analysis software. **However, a disadvange is that the audio may be played back at a low sound level by many audio players.** You may need to adjust a gain factor to play back audio at a comfortable level.
- Stored audio files take up considerable amounts of storage space (about 700 MB per hour). This fills up the storage capacity of the device quickly.
- On a related note, keep in mind that downloading large amounts of recorded audio data to a PC over the USB2.0 interface will take some time.

14.6. Downloading screenshots

Firmware Every time you press the screenshot button (in the top bar), a copy of the current screen in .PNG format is saved to the folder SCRNSHOT. A serial number is added to each file. Copy the screenshots to your PC from this folder.

14.7. Further use of USB

Firmware updates are also done by connecting the AM100 via USB. Instructions on how to perform firmware updates are given in the chapter 16 of this manual.

15. Remote control

In certain situations you may wish to control your Bedrock AM100 remotely from a PC, smartphone or tablet. To achieve this, there are two ways: our Bedrock Studio application and our Remote Control API. Either way, a connection needs to be established first.

15.1. Establishing a connection

There are three alternative ways to establish a connection:

- **USB**: Connect a USB-C cable to your PC.
- **Bluetooth:**Press "pair" from the top bar settings menu (gear icon). The Bedrock AM100 is now discoverable for other devices; the Bluetooth icon in the top bar colors blue. Now confirm the Bluetooth connection from the host device to which you wish to connect the AM100. Once a connection is established, the Bluetooth icon turns yellow. If the connection becomes active it turns white.
- **WiFi:** connect your AM100 to a local WLAN from the settings module (see chapter 12 for detailed instructions). You can now connect from any other device within the same WLAN. You will need to know the IP address for the AM100 for that, which you find in the Settings module.

When connected to a WLAN, the AM100 is only seen by devices within the same network. No connections are made to external services or to the internet. If you need to control the AM100 from a remote location outside the local WLAN, you need set up port forwarding and NAT. Consult the manual of your network router for the configuration procedure.

Include the following settings in your network configuration:

- Set up the DHCP server to associate a fixed local IP number to the AM100.
- Set up port forwarding to this IP number for port 23. Connect to an external port number of your own choosing.
- From the internet, connect to the public/external IP address of your router and the port number you chose.
- Now set up the remote connection in the same way you would within a local WLAN.

Note that port forwarding may have implications for the security of your network. Where applicable, consult your network administrator before making network configuration changes. We recommend disabling port forwarding if external connections to the AM100 are not currently used.

15.2. Remote duplication of the AM100 screen (Bedrock Studio)

The easiest way to remotely control a Bedrock AM100 is by cloning the screen to a difference device using our Remote Display application. The remote display application does exactly what the name says: you will get the exact same controls that you have on the Bedrock AM100 device itself, but the entire display is simply shown on a different device.

We currently offer a Windows 10 application only. We plan to release an Android app around March 2022. These applications can be downloaded from the Bedrock Audio website.

> ⚠️ The Bedrock Studio application needs to have the exact same firmware version that your Bedrock AM100 is running. If your AM100 is connected via USB, the correct firmware is automatically copied from the AM100 to your host device. On Bluetooth and WiFi connectections, the Remote Display application may ask you to open the correct .EAP firmware package on your local computer, or automatically download it from the internet.

How you set up the connection from the Windows application to the AM100 depends on the type of connection. Choose how you wish to connect (USB/Bluetooth/Wifi) and then configure the connection accordingly.

- **USB**: The connection is established through a virtual COM port. If you know the COM port number, you can enter this manually. Alternatively, the application can be instructed to scan for AM100 devices connected to USB.
- **Bluetooth:** Once the AM100 is paired over Bluetooth,the connection to the application is made automatically.
- **WiFi:** The application needs to be provided with the IP address, port number and access code for the AM100. To obtain this information, go to the "WiFi and network" tab in the settings module on the AM100.

Once the connection is set up, you should see screen of the AM100 appear on your host device. Now you can operate the AM100 from the host device in the exact same way as if you were working on the device locally. Note that all files and results are stored on the AM100 itself, and not on the host device. Copying the measurement results to the host device is a separate operation (see chapter 13 of this manual).

All functions on touch screen on the AM100 are temporarily disabled as soon as a connection to the Remote Display application is made. Touch the display to return control to the Bedrock AM100. If you are connecting via USB, always make sure that you are using high-quality USB cables, especially when operating with a long cable between the Bedrock AM100 and your PC. Most USB connection issues are traced back to poor quality cables.

The resolution of the display is 480x800 pixels. You can stretch the window of the Remote Display application to enlarge it. It may be convenient to use an enlarged version of the display (e.g. in classroom settings), but while increasing the size, the resolution remains 400x800.. It is therefore normal for the enlarged display to look somewhat blurry.

For further documentation on Bedrock Studio application currently serves mostly as a tool for remote display cloning as described above. We plan to add many new features in the upcoming period. For the most up-to-date information, consult the documentation for Bedrock Studio from our website.

15.3. The Bedrock remote control API

Each Bedrock AM100 device that is connected to USB attempts to set up a USB-CDC serial connection with its host. This allows you to connect to the AM100 from your own software.

Our remote control API is available upon request, free of charge, for every user of a Bedrock AM100 device. A full description of the API is well beyond the scope of this manual. We will briefly describe the characteristics of the API here. For sample code and documentation, please contact support@bedrock-audio.com.

Through serial commands, you can control virtually every function and control that the user also has control over via the display. You can launch and terminate modules, set parameters and start and stop measurements at will. Measurement results can be read by a host application, or results can be saved on the AM100 and then retrieved by accessing the AM100 as a mass storage device.

Since all calls are implemented as commands sent over a serial connection, in principle the AM100 could be controlled via a simple terminal program. A list of serial commands is provided as part of the API. Using these commands, remote control can be implemented on any platform, in any programming language, as long as the host device implements USB-CDC.

Most commands (such as commands for starting and stopping measurements, or requesting software versions) are the same across all modules. On top of these generic commands there are module-specific commands for each module.

To further simplify the development of dedicated control software by the end user a C# library is provided that implements the serial commands.

16. Power and charging

16.1. Charging instructions

The Bedrock AM100 can be charged with any USB charger capable of a charge current of 1500 mA or higher, such as the one originally supplied with your AM100.. Some PCs allow increased charge currents to be drawn from their USB ports (sometimes through special ports marked "Boost" or "Charge" or with a lighning bolt next to the USB logo). These can then also be used to charge the instrument. If a USB Power Delivery 2.0 (or higher) fast charger is used, this will minimize the charging time.

⚠️ Note: a standard USB port on a PC or Mac only provides a limited current. This current may be insufficient to simultaneously charge and operate the device. Therefore, it is possible that the battery is slowly drained while the AM100 is connected to a PC. The battery icon is animated while charging. If the battery icon does not show an animated (moving) bar, the AM100 is not being charged.

The Bedrock AM100 has internal charge circuitry that optimizes your charge time while safeguarding the life span of your battery pack. Also if you are not using the device for a prolonged period, you can safely keep it connected to a charger.

A few charging tips:

- A LED next to the USB shows the charge status. While blinking (alternating between green and amber), the battery is still charging. The LED burns constant green as soon as the battery is full.
- If you find, over time, that the battery capacity is reduced significantly compared to its original capacity, we recommend that you fully discharge the batteries, and then fully recharge them. If necessary, this procedure may be repeated a few times.
- If the battery charge indicator appears to become inaccurate (e.g., the battery is drained quickly from the point that the indicator still shows 50% charge), it also helps to completely discharge and recharge the battery.
- It is normal for the total battery capacity to slowly decrease over time. After 500 charge/recharge cycles, the battery capacity is normally at 80% to 90% of its original capacity, depending on the way the device is used and recharged.
- If you start using the AM100 after a long period of storage, we recommend that you first top off the charge before using the instrument. You may also want to check if the date and time settings were preserved.

- Unlike the Bedrock SMxx models, the AM100 can be operated while being charged at the same time.

A lightning symbol is displayed to the left of the battery icon. The battery icon provides information on the power and charging status in the following ways:

- If the battery icon is animated, the battery is being charged. The LED also blinks green/amber in this state.
- If the lightning bolt is yellow and pointing downwards, the device is charging. The bigger the lighning symbol, the faster the device is charging. This depends on the capabilities of your charger or your computer's USB port.
- If the lightning bolt is purple and and pointing upward, then the AM100 is providing power to an accessory through USB.
- If the lightning bolt is white, the device is being powered through USB but not trying to charge (the battery is already full or nearly full).

16.2. Power saving recommendations

How long the device can be used on a single charge depends on the mode and intensity of use. Power conservation settings can be controlled from the device's "hardware settings" menu. With its default settings, approximately 350 STIPA measurements can be carried out over the course of an 8 hour day without recharging.

Some ways to extend the battery life:

- Switch off the device when you do not plan to use it for a while
- Choose conservative power settings in the hardware menu. The LCD backlight is a considerable factor in the overall drain on the battery.
- Disable "pre-amp phantom power" if you use a microphone that does not need it.

16.3. Use of power banks and external battery packs

For very long measurement sessions, the battery capacity of the internal battery may be insufficient. If you cannot spare the time to recharge in the middle of your session, we recommend that you use an external USB battery pack (aka power bank). Even though the AM100 has been certified to comply with applicable EMC standards, also when connected to an AC source, the signal quality is always slightly better when powered from batteries.

When you connect the AM100 to an external battery pack, a dialog will appear asking you whether you want to charge the device or operate on external power. We recommend that you do not recharge the internal battery from the power bank, so choose "external." The internal battery will then not be charged, nor discharged, while measuring. **Note that the battery pack should have an output current rating of at least 1500 mA.**

17. Cleaning, maintenance and calibration

With normal use, no further maintenance is needed beyond normal cleaning periodic re-calibration of the device. We recommend that you have the device checked and calibrated at our factory or at an external calibration specialist at least once every 2 years.

We recommend having the battery pack replaced after approx. 500 charge/recharge cycles, or once every 4 years.

We recommend that you clean the instrument with a soft cloth. LCD display cleaner can be used to clean the touch screen, provided that the following precautions are taken:

• Do not spray cleaner directly onto the display
• Only use cleaning liquids specifically intended for LCD screens

Never use aggressive cleaning agents or solvents (such as alcohol or white spirit) to clean the AM100. The surface coating on the plastic parts of the device will be damaged. Also take care not to handle the AM100 if your hands are still wet from using hand alcohol or lotion.

Battery disposal

The device contains a 4.8v rechargeable NiMH battery pack. At the end of the life cycle of this product, care should be taken to dispose of this battery pack responsibly, in compliance with rules and regulations for recycling of batteries that apply in your region. MAKE SURE THAT THIS BATTERY PACK IS REMOVED BEFORE DISPOSING OF THE DEVICE. If you are unable to remove the battery pack yourself (by unscrewing the back lid of the device), make sure that you mention the presence of an NiMH battery pack when handing in the entire device for recycling.

18. Product warranty

The device is covered by a full-service, worldwide pick-up & return warranty until 24 months after the date of purchase. Please direct all warranty claims to:

> Bedrock Audio BV
> Ampèreweg 18
> 2627 BG DELFT
> THE NETHERLANDS
> warranty@bedrock-audio.com

Please email us and wait for our response (typically within a business day) before shipping your hardware back. We will ask you to fill out an RMA form, and we will supply you with a reference number for your warranty claim. If you can provide a complete statement of the problems you are experiencing, this will help us solve your hardware issues as soon as possible. Always provide the ID number of your device (found on the bar code label on the back of the device), your name and address, and your date of purchase. If you did not purchase directly from the Bedrock webshop, also include dated proof of your purchase.

In general, our warranty programme covers all defects except for those resulting from accidents, misuse (including improper electrical connections) and improper maintenance. The following is explicitly excluded from product warranty:

- Any damage resulting from dropping, falling or excess vibration.
- Any damage done by liquids, including damage resulting from excessive use of screen cleaning liquids or aggressive cleaning agents or solvents.
- Any damage resulting from incorrect electric connections to the device through the USB and/or audio connectors.
- Any damage done by power surges or overvoltage on the USB power supply and/or audio connector.
- Any mechanical damages to the display caused by mechanical forces applied to the screen surface, such as scratches and cracks.
- Any damage resulting from attempted changes to the system firmware, included changes to the files on the system partition of the device.
- Normal wear and tear.

The following voids warranty:

- Repair attempts by an unauthorized party
- Replacement of battery packs by an unauthorized party
- Removal of product stickers, tamper proof seals, bar codes or serial numbers
- Attempts to install firmware from any other source than the manufacturer, or firmware intended for a different device model.

Malfunctions that are found to result from any of the above conditions are not covered by warranty. Repairs will take place only at the expense of the owner. If claimed under warranty, an examination fee may be charged and transportation costs will be charged to the owner.

Please do not attempt to open your Bedrock AM100 yourself, as there are no user-serviceable parts inside. If you do see a legitimate need to open the device yourself, please contact warranty@bedrock-audio first, so we can discuss preserving your warranty rights. Please be aware that the product is protected by external and internal tamper-proof seals.

A worldwide pick-up&return service is included in the warranty. This means that we will have the defective unit picked up at your location at no charge, provided that it is presented in its original equipment case, packed in a cardboard box with suitable cushioning material.

Please note that Bedrock Audio and Embedded Acoustics explicitly reject liability for any damages resulting from incorrect measurement results obtained while using any of our products. Ensuring that measurement results are valid is the responsibility of the end user of the instrument.

19. Firmware updates

19.1.Getting the latest firmware for your device

Firmware updates are released through our website Bedrock-audio.com. Before updating your firmware, always check:

- If the update you downloaded is intended for your device type (AM100)
- If you have sufficient battery charge to complete the update cycle without interruption

19.2. Installing new firmware

Executing a firmware update is easy:

- Choose and download the correct firmware update package from www.bedrock-audio.com/support
- Connect the device to any PC or Mac through USB
- When prompted on the LCD screen, choose "connect through USB"
- The AM100's internal file system can now be accessed from your computer in the same way as any mass storage device
- Copy the firmware updated package to the folder named "UPDATES"
- Now restart the device using its power button

The firmware update procedure will automatically start. You will be able to see the progress of the update procedure on the LCD display. The update may take up to 15 minutes, depending on the size of the update package.

Interrupting the power supply during the update process may render your device unusable. If this happens, consult secion 18.3 (recovery mode).

⚠ Never interrupt the firmware update procedure. Make sure that the device is fully charged or connected to a stable power source before starting the update procedure. Please be aware that the update procedure may take a significant amount of time, depending on the update package that you are installing.

⚠ Never try to install an update package intended for a different device model. The update package will most likely not install; if it does, it will render your device inoperable. Damage sustained through attempts to install incorrect firmware update packages is not covered by warranty.

19.3. Recovery mode

If the internal operating system or software of the Bedrock AM100 gets corrupted, because of a failed firmware update or otherwise, the system can usually be restored through the internal recovery mode.

To enter recovery mode, make sure the device is turned off (shutdown can be forced by keeping the on/off button pressed for >5 seconds).

Then take the following steps:

- Start the device by pressing the power button. Press the button briefly and release, until the LED lights up.
- After a very short interval (<1 second) the LED turns green. As soon as this happens, press the power button again and keep it pressed down.
- Timing is critical; it may take a few attempts to get it right. It is a short press to start the device, followed by a long press as soon as the LED turns green.
- Wait for the AM100 to initiate recovery mode. You will see the screen below.

Entering recovery mode
USB mass storage will be connected
Press power button to reboot
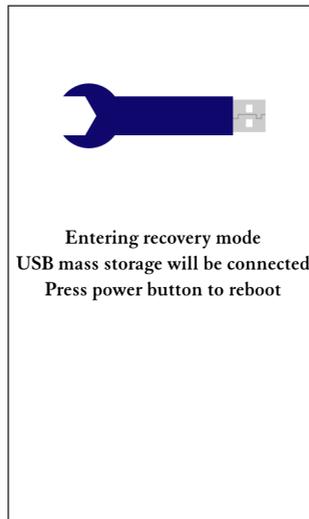
The device can now be accessed through USB. This means you can copy a new .EAP update package to the UPDATES folder. If you are having trouble accessing USB in the normal operating modes of the AM100, this may be a way to restore your system and re-install the latest firmware.

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⚠️ In order for recovery mode to work, some core system files are needed. If you accidentally erased the entire system partition or if the system partition has become completely corrupted, then recovery mode will not work work. In these rare cases, the device needs to be returned to the manufacturer for recovery.

20. Buttons, icons and symbols

Common buttons, icons and symbols shared between modules	
Home	Home Sends you back to the main menu of the SM50
start / stop	Start /stop measurement
save / save	Save button Disabled if grey (no data to save)
(progress meter)	Progress meter. Shows progress of current measurement (fully green = finished)
●	Measurement running Showing while recording is running
48V	This is showing if phantom power is enabled in "settings" Turns from grey to black is phantom power is active
◠◠	Audio recording indicator Showing if audio recording is enabled (NB: tap "save" to store)
50%	Remaining battery charge
charge	Battery charging
ext.power	Operating on external power
#0001	Measurement number Serial number of current measurement
(busy icon)	Instrument busy Occurs when switching modules
browse measurements	Menu navigation. Tap arrows to flip through menus (red block indicates position of current menu within menu structure).
previous / next	Browse forward / backward through stored measurements
note	Text annotation stored for this measurement. Press to read

Start delay	
	An animated clock dials counts down until the start of the measurement. Set the countdown time in the Settings module; choose 0 seconds as a setting to start each measurement immediately upon pressing the "start" button

Settings module	
	Set system clock First adjust the date and time; then tap this button to confirm
	Calibrate touch screen Tap this button to initiate touch calibration procedure
Quick STIPA / STIPA pro module	
U E A+	Qualification label U-A as defined in IEC-60268-16 (derived directly from STI)
N!	STI shown includes the effect of "additive noise" according to the noise spectrum entered in the "additve noise" menu
▲ / ▼	Adjust graph scale up / down
SPL meter module	
	Overload indicator
	Underrange indicator
set...	Set time and frequency weightings. Settings keyboard appears upon tapping this button

RTA meter module	
move cursor	Move cursor through frequency bands. Tap left to move to lower frequencies, right to move to higher frequencies.
FFT module	
▶◀ hide ⊕Q	Zoom controls, which appear if the magnifying glass icon on the chart is pressed. Zoom in (+) or out (-), and center around the current cursor position. Tap "hide" to collapse.

RT60 module	
● arm	Press button to arm the instrument. The measurement is then automatically triggered once the energy level drops.
accept individual #1	Press to accept the current individual measurement, and to incorporate data from this measurement into the overall mean.
> EDT T20 T30	Select which reverberation metric to display (EDT, T20 or T30). Press multiple times to cycle through the options.
✓	The reverberation time measured in this frequency band is consistent and reliable. Measurement error $<= 0.1s$
⚠	The reverberation time measured in this frequency band may not be fully reliable; the signal-to-noise ratio is less than required, or the energy decay curve is not be sufficiently linear. The result will be indicative of the true reverberation time, but the measurement accuracy is not guaranteed.
✖	On theoretical grounds, or based on the measurement data, the reverberation time measured in this frequency band is considered unreliable.

LAEQ logging module	
Average LAEQ below limit	The A-weighted equivalent-continuous sound level (LAEQ) is below the preset limit. This is true when considering the current (short) period (<1min) as well as when considering the running average (typically 15 minutes).
Current LAEQ above limit	The running average is still below the limit, but the current level exceeds the limit. The running average will soon exceed the limit as well, unless the sound level is reduced in the upcoming period.
Average LAEQ above limit	The running LAEQ average exceeds the limit.
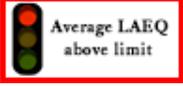
USB audio mode	
disconnected	USB is not connected.
connected	USB is physically connected. A data connection with Windows audio drivers may or may not have been established
data link inactive	The USB data link is inactive. No audio is streaming
data link active	The USB data link is active and audio is being streamed digitally to your computer
data link error	An error occurred while setting up the data link

NOTE: this table does not describe all buttons and functions. Generally, buttons will be marked with captions that explain their function.

21. Technical specifications

Functionality	Speech Transmission Index (STIPA and Full STI) Speech Level Meter (IEC-60268-16) Sound Pressure Level (A,C, Z, slow, fast, impulse, Leq, SEL, pk) Statistical Acoustics (percentiles) 1/1, 1/3, 1/6 and 1/12 octave RTA RTA and SPL logging FFT analyzer RT60 reverb time measurements (interrupted noise and impulse) Noise curves (NR, NC, PNC, NCB, RC, RCmarkII) Oscilloscope, Loudspeaker Polarity test, Frequency Estimation, THD+N, AC volt meter (Vrms, dBU, dBV) Sound insulation, SRI/STC USB audio device mode Remote operation via USB
Standards compliance	IEC 60268-16 rev. 4, IEC 61260 (class 1), ISO 3382-2, ANSI S12.2, ISO-1996, IEC-61672 Group X Class 1; ANSI S1.4 Type 1
RTA filter specifications (IEC-61260)	Class 1 1/1, 1/3, 1/6 and 1/12 octave filters (base 10)
Acoustic specs device	Effective noise floor: < 14 dB(A) (LO range, 20 Hz - 20 kHz) Upper limit peak range: 128 dB (HI range, 20 Hz - 20 kHz) Linear range: See separate table below Maximum permissible sound pressure level is 155 dB Maximum electric input signal is 30 Vpp
	Microphone type BAMT1-MP201, 48V phantom powered 1/2" omnidirectional free-field microphone Detachable IEC 61094-4 Type WS rear-vented mic capsule Capsule housing alloy, diaphragm nickel Conforms with class 1 / type 1 requirements Capacitance 16 pF 50 mV/Pa sensitivity Dynamic range (3% distortion limit) > 146 dB Temperature Coefficient: -0.012 dB/°C Pressure Coefficient (250 Hz): - 0.004 dB/kPa
Display	Capacitive touch, TFT display 800x480 pixels, 16.7M colors Luminance typ. 400 Cd/m2 Viewing angle 70 deg Typ. lifetime 20,000 h

Data storage	30 GB internal storage for user data plus 2 GB system partition Accessible from PC through USB (mass storage mode)
Environmental	Temperature: 5 -35°C (41-95F) Relative humidity: 0% - 95% (non-condensing)
Power	NiMH battery pack 4.8V nominal, typ. 2500 mAh Typical battery lifetime >4h Max. charge current 1.8A
Start-up period	The AM100 is ready for measurements 8 seconds after powering up
Overload and under-range indicators	Overload and underrange conditions are indicated with the resp. icons as shown in section 17 of this manual. Icons will show when levels outside the linear range are measured (See table below).
Electromagnetic compatibility	Complies with EN 61326-1:2006 - EN 61326-1:2006 FCC part 15 compliant (class A device)
Mechanical specs	Size approx. 210 x 85 x 55 mm (excl. microphone) Weight 600 g (including microphone) Rear-mounted tripod adapter Hook for carrying cord
Bundled items	Carrying case, measuring instrument, microphone, USB charger, int. charger adapters (US, EU, UK, AU), USB charging/data cable, manual, carrying cord, USB flash drive with test signals and software tools.

Linear range (31 Hz - 4k - 8k - 12.5kHz)

Upper and lower limits of linear range (dB SPL)	Low range			High range Reference range		
	A	C	Z	A	C	Z
BAMT1 - MP201**	24-*	24-*	24-108	31-*	31-*	31-124

Upper and lower limits of linear range (dBuV)	Low range			High range Reference range		
	A	C	Z	A	C	Z
Electrical input (XLR)	20-*	20-*	21-122	37-*	35-*	38-133

*The overload for A and C-weighting is dependent on the frequency of the applied test signal. To determine the overload value, correct the Z-weighted overload with the appropriate weighting.E.g., for 8 kHz, correct the Z-weighted upper limit with -1.1 dB for A-weighting and -3.0 dB for C-weighting (per the definitions of A and C weightings).

** Linear range specifications determined for BAMT1-MP201 microphones with a specified sensitivity of 50 +/- 2 mV/Pa

22. Troubleshooting

Problem	Possible cause	Solution
System does not power on after pressing power button	Battery empty	Connect wall charger for at least 30 minutes
System does not recharge after connecting through USB	Charge current too low	Use bundled wall charger and USB cable; standard PC USB ports do not supply enough current to charge
Battery lifetime is too short	Battery not fully charged Battery is worn	Keep device on charger until it completes its charge cycle. Have battery replaced after 500 charge cycles or 4 years.
Battery indicator is inaccurate	Battery charge measurement system poorly calibrated (e.g. due to self-discharge after prolonged shelving)	Completely drain the battery, and then fully recharge. Repeat if necessary.
Device warms up during use	Lack of ventilation	Warming up of the device to temperatures of approx. 35C/95F is normal. If the device runs considerably hotter than that, check ventilation and contact support if the problem persists
Device not recognized by PC when connected through USB	Driver problem Device in measurement mode	Update PC drivers (SM50 is approached as a mass storage device) Select USB data transfer mode when dialog window appears on the SM50 screen
Display backlight turns off, no response to touch screen	Backlight time-out event	Adjust backlight time-out in power saving settings. Pressing power button turns the display back on
Device "hangs," becomes unresponsive, or will not power up after resetting, and/or LED next to USB port burns red.	Internal software problem	Keep the power button pressed for 10s. This forces the device to shut down. Then press the power button again to restart. If the problem persists, contact support

LED states

LED state	Meaning
Burning green	Device operating
Blinking amber/green	Device charging
White	Hardware failure. Contact support.
Burning or blinking red	Software crash, hard reset needed (keep power button pressed 10s)
Blue	Wireless connection active (BT/WiFi)
Magenta	USB-C port is powering an accessory

Firmware update error codes

LED state	Meaning
Error 17	**Not a valid file.** The update package failed to produce the correct checksum during the file verification procedure. This is usually caused by a corrupted or incomplete package download. If this error persists on Windows 7, try this: - Find the drive letter for the AM100, and right click. Choose "properties" - Go to the hardware tab. Click on "properties" - Click "change settings" - Go to the "policies" tab - Select "Better performance" instead of "Quick removal"
Error 18	**File transfer failure.** The transfer of one of the packed files failed. This may have various causes, but most likely it is a (possibly temporary) issue with the internal file system. If you keep getting this error, please contact support.
Error 19	**Update package does not match device type.** The update package copied to your device is intended for a different model. Please download an update package for your device model and try again.
Error 20	**The update package does not contain device type info.** The update package copied to your device does not contain device type information. Most likely, you are trying to install an older firmware version on a newer device. Try installing a package for version 3.0.0 or higher. If you urgently need to roll back to an older firmware version, contact support.

Annex A. Correction factors for SPL measurements (IEC-61672)

All of the microphone types distributed with the AM100 are designed as free-field microphones. The BAMT1 modules feature detachable 1/2" capsules. In this annex, data are provided that may be needed in any of the following cases:

- **In case you are an end user who wishes to manually correct measurements for maximum accuracy.** End users of the Bedrock AM100 who aim to achieve the highest possible accuracy may correct free field measurements with the (typical) free-field response of the applied microphone. Similarly, they may wish to correct for the frequency-dependent effect of the windscreen or for reflections against the instrument body.
- **In case you are a test lab and need to perform periodic tests according to IEC-61672-3.** This standard defines the procedures for periodic testing of acoustic measuring instruments, in particular when used for measuring sound pressure levels. IEC-61672-3 requires the free field response to fall within certain tolerance limits, after application of correction factors provided by the manufacturer. Those correction factors are given in this Annex for each type of microphone offered with Bedrock instruments.

The following microphone data are provided for all microphone types that are available with the Bedrock AM100:

- **Typical free field response.** Since our microphones are free-field microphones, the free field response should ideally be zero for all frequencies, but in practice small deviations occur, particular at higher frequencies.
- **Typical actuator response.** Calibration labs that use electrostatic actuators to verify the microphone response need the free-field to actuator corrections specified in this Annex.
- **Typical pressure field response.** Test labs that use a the multi-frequency calibrator for periodic testing (such as the B&K4226) need to apply the free-field to pressure field corrections specified in this Annex.
- **Effects of reflections from the instrument body.** Reflected sound from the instrument body have a small but measurable impact on the measured sound field. The overall response, including these effects, needs to fall within the tolerance limits of IEC-61672-3 during periodic testing.
- **Effects of wind screens.** The 32mm wind screens supplied with the BAMT1 and BAMT2 microphones have a measurable impact on the microphone response. If these windscreens are included in the scope of periodic testing, the overall response (including wind screens) needs to fall within the tolerance limits.

A1. BAMT1-MP201 microphone

The BAMT1-MP201 is the default microphone-preamp combination bundled with the Bedrock AM100. The MP201 is a high-precision Class I / Type I free field microphone capsule.

A1.1. Typical free field response and actuator to free field corrections for the BAMT1-MP201.

The typical free field response of the MP201 capsule is specified in this section. The MP201 (including grid) was measured by the manufacturer at 0 degrees incidence, based on average values of microphone capsules against a B&K 4180 response using a B&K 4228 Pistonphone. Microphones were mounted on M201 preamp. The typical actuator reponse was determined using a GRAS RA0015 and a GRAS 14AA. The measurement uncertainty is +/- 0.2 dB.

We specify three series of measurements results in this paragraph, marked as columns (1), (2) en (3) in the table below. A description of the data in these columns, and their uses for measurement and calibration, is given below.

- Column (1) is the typical free-field response of the microphone. To enhance the accuracy of free-field measurements, users may subtract the typical free field response values in dB from the measured sound pressure levels per frequency band.
- The typical free field response of column (1) is also used for periodic electric testing according to 61672-3. In that case, the typical free field response (1) is added to the measurement values to obtain an equivalent acoustic sound pressure level when the capsule is removed and an electrical test signal is applied at the microphone preamplifier input using a 16 pF coupler.
- The typical actuator response of column (2) is not usually applied directly, but was measured to be able to calculate the actuator to free-field corractions of column (3)
- For periodic acoustic or actuator testing according to 61672-3, the actuator-to-free-field correction of column (3) is added to the measurement values to obtain an equivalent acoustic sound pressure level when the capsule is measured using an electrostatic actuator. The actuator-to-free-field is in literature also often referred to as the free field correction.

BAMT1-MP201 Frequency	(1) Typical free field response	(2) Typical actuator response	(3) Actuator to free field correction
20 Hz	0.0	0.0	0.0
25 Hz	0.0	0.0	0.0
31.5 Hz	0.0	0.0	0.0
40 Hz	0.0	0.0	0.0
50 Hz	0.0	0.0	0.0
63 Hz	0.0	0.0	0.0
80 Hz	0.0	0.0	0.0
100 Hz	0.0	0.0	0.0
125 Hz	0.0	0.0	0.0
160 Hz	0.0	0.0	0.0
200 Hz	0.0	0.0	0.0
250 Hz	0.0	0.0	0.0
315 Hz	0.0	0.0	0.0
400 Hz	0.0	0.0	0.0
500 Hz	0.0	0.0	0.0
630 Hz	0.0	0.0	0.0
800 Hz	0.1	0.0	0.1
1 kHz	0.2	0.0	0.2
1.25 kHz	0.2	-0.1	0.3
1.6 kHz	0.3	-0.1	0.4
2 kHz	0.3	-0.2	0.5
2.5 kHz	0.3	-0.3	0.6
3.15 kHz	0.3	-0.5	0.8
4 kHz	0.3	-0.7	1.0
5 kHz	0.4	-1.1	1.5
6.3 kHz	0.4	-1.8	2.2
8 kHz	0.4	-2.9	3.3
10 kHz	0.4	-4.2	4.6
12.5 kHz	0.5	-6.0	6.5
16 kHz	-0.2	-8.2	8.0
20 kHz	-2.2	-11.2	9.0

A1.2. Typical pressure field to free field corrections for the BAMT1-MP201

The correction factors below are added to the measured values to arrive at the expected free field sound pressure level measurement when using the B&K4226 multi-frequency calibrator.

Frequency	BAMT1-MP201
31.5 Hz	0.3
63 Hz	-0.1
125 Hz	-0.1
250 Hz	-0.1
500 Hz	-0.1
1 kHz	-0.06
2 kHz	0.1
4 kHz	1.0
8 kHz	2.5
16 kHz	5.0

A1.3. Reflections from the instrument body when using the BAMT1-MP201

If the microphone is fitted directly on top of the AM100, without any length of XLR cable inbetween, then reflections from the instrument body will affect measurements. The table below gives the effect of those reflections per octave band. To avoid these reflections, use a cable with a minimum length of 1 meter. As a user, subtract the values from the table below to correct readings. To apply corrections during periodic testing, add these same values.

Frequency	BAMT1-MP201
31.5 Hz	-0.02
63 Hz	0.00
125 Hz	0.00
250 Hz	0.02
500 Hz	0.03
1 kHz	-0.01
2 kHz	0.02
4 kHz	0.01
8 kHz	0.04
16 kHz	-0.01

A1.4. Corrections of the 32mm windscreen with the BAMT1-MP201

A 32mm-1/2" windscreen is supplied standard with each AM100 fitted with a BAMT1 microphone. The effect of this windscreen per octave band is given in the table below. As a user, subtract the values from the table below to correct readings. To apply corrections during periodic testing, add these same values.

Frequency	Windscreen effect
31.5 Hz	0.0
40 Hz	0.0
50 Hz	0.0
63 Hz	0.0
80 Hz	0.0
100 Hz	0.0
125 Hz	0.0
160 Hz	0.0
200 Hz	0.0
250 Hz	0.0
315 Hz	0.0
400 Hz	0.0
500 Hz	0.0
630 Hz	0.0
800 Hz	0.1
1 kHz	0.1
1.25 kHz	0.1
1.6 kHz	0.1
2 kHz	0.2
2.5 kHz	0.2
3.15 kHz	0.3
4 kHz	0.5
5 kHz	0.6
6.3 kHz	0.8
8 kHz	0.7
10 kHz	0.3
12.5 kHz	-0.2
16 kHz	-0.5
20 kHz	-0.4

A2. BAMT1-MP309 microphone

The BAMT1-MP309 is an alternative class 2 microphone-preamp combination offered for use with the Bedrock AM100. The MP309 is a high-end Class II / Type II free field microphone capsule.

A2.1. Typical free field response and actuator to free field corrections for the BAMT1-MP309.

The typical free field response of the MP201 capsule is specified in this section. The MP201 (including grid) was measured by the manufacturer at 0 degrees incidence, based on average values of microphone capsules against a B&K 4180 response using a B&K 4228 Pistonphone. Microphones were mounted on M201 preamp. The typical actuator reponse was determined using a GRAS RA0015 and a GRAS 14AA. The measurement uncertainty is +/- 0.2 dB.

We specify three series of measurements results in this paragraph, marked as columns (1), (2) en (3) in the table below. A description of the data in these columns, and their uses for measurement and calibration, is given below.

- Column (1) is the typical free-field response of the microphone. To enhance the accuracy of free-field measurements, users may subtract the typical free field response values in dB from the measured sound pressure levels per frequency band.
- The typical free field response of column (1) is also used for periodic electric testing according to 61672-3. In that case, the typical free field response (1) is added to the measurement values to obtain an equivalent acoustic sound pressure level when the capsule is removed and an electrical test signal is applied at the microphone preamplifier input using a 16 pF coupler.
- The typical actuator response of column (2) is not usually applied directly, but was measured to be able to calculate the actuator to free-field corractions of column (3)
- For periodic acoustic or actuator testing according to 61672-3, the actuator-to-free-field correction of column (3) is added to the measurement values to obtain an equivalent acoustic sound pressure level when the capsule is measured using an electrostatic actuator. The actuator-to-free-field is in literature also often referred to as the free field correction.

BAMT1-MP309 Frequency	(1) Typical free field response	(2) Typical actuator response	(3) Actuator to free field correction
20 Hz	0.0	0.0	0.0
25 Hz	0.0	0.0	0.0
31.5 Hz	0.0	0.0	0.0
40 Hz	0.0	0.0	0.0
50 Hz	0.0	0.0	0.0
63 Hz	0.0	0.0	0.0
80 Hz	0.0	0.0	0.0
100 Hz	0.0	0.0	0.0
125 Hz	0.0	0.0	0.0
160 Hz	0.0	0.0	0.0
200 Hz	0.0	0.0	0.0
250 Hz	0.0	0.0	0.0
315 Hz	0.0	0.0	0.0
400 Hz	0.0	0.0	0.0
500 Hz	0.0	0.0	0.0
630 Hz	0.0	0.0	0.0
800 Hz	0.1	0.0	0.1
1 kHz	0.1	-0.1	0.2
1.25 kHz	0.2	-0.1	0.3
1.6 kHz	0.2	-0.2	0.4
2 kHz	0.2	-0.3	0.5
2.5 kHz	0.2	-0.4	0.6
3.15 kHz	0.2	-0.6	0.8
4 kHz	0.1	-0.9	1.0
5 kHz	0.1	-1.4	1.5
6.3 kHz	0.2	-2.0	2.2
8 kHz	0.4	-2.9	3.3
10 kHz	0.6	-4.0	4.6
12.5 kHz	1.2	-5.3	6.5
16 kHz	1.1	-6.9	8.0
20 kHz	-0.4	-9.4	9.0

A2.2. Typical pressure field to free field corrections for the BAMT1-MP309

The correction factors below are added to the measured values to arrive at the expected free field sound pressure level measurement when using the B&K4226 multi-frequency calibrator.

Frequency	BAMT1-MP309
31.5 Hz	0.3
63 Hz	-0.1
125 Hz	-0.1
250 Hz	-0.1
500 Hz	-0.1
1 kHz	-0.06
2 kHz	-0
4 kHz	0.5
8 kHz	1.8
16 kHz	4.8

A2.3. Reflections from the instrument body when using the BAMT1-MP309

If the microphone is fitted directly on top of the AM100, without any length of XLR cable inbetween, then reflections from the instrument body will affect measurements. The table below gives the effect of those reflections per octave band. To avoid these reflections, use a cable with a minimum length of 1 meter. As a user, subtract the values from the table below to correct readings. To apply corrections during periodic testing, add these same values.

Frequency	BAMT1-MP309
31.5 Hz	-0.02
63 Hz	0.00
125 Hz	0.00
250 Hz	0.02
500 Hz	0.03
1 kHz	-0.01
2 kHz	0.02
4 kHz	0.01
8 kHz	0.04
16 kHz	-0.01

A2.4. Corrections for the 32mm windscreen with the BAMT1-MP309

A 32mm-1/2" windscreen is supplied standard with each AM100 fitted with a BAMT1 microphone. The effect of this windscreen per octave band is given in the table below. As a user, subtract the values from the table below to correct readings. To apply corrections during periodic testing, add these same values.

Frequency	Windscreen effect
31.5 Hz	0.0
40 Hz	0.0
50 Hz	0.0
63 Hz	0.0
80 Hz	0.0
100 Hz	0.0
125 Hz	0.0
160 Hz	0.0
200 Hz	0.0
250 Hz	0.0
315 Hz	0.0
400 Hz	0.0
500 Hz	0.0
630 Hz	0.0
800 Hz	0.1
1 kHz	0.1
1.25 kHz	0.1
1.6 kHz	0.1
2 kHz	0.2
2.5 kHz	0.2
3.15 kHz	0.3
4 kHz	0.5
5 kHz	0.6
6.3 kHz	0.8
8 kHz	0.7
10 kHz	0.3
12.5 kHz	-0.2
16 kHz	-0.5
20 kHz	-0.4

A3. BAMT2 microphone

The BAMT2 is the standard microphone-preamp combination bundled with the Bedrock SM30 and SM50 The BAMT2 is a highly stable 1/4" Class II / Type II electret microphone.

A3.1. Typical pressure field to free field corrections for the BAMT2

The correction factors below are added to the measured values to arrive at the expected free field sound pressure level measurement when using the B&K4226 multi-frequency calibrator. Note that the a different set of corrections applies to the 1/4" adapter of the B&K4226 (type designation DP0781) and Bedrock 1/4" adapter (type designation CALADPT, which also comes bundled with the BAC2 calibrator). Both sets of correction values are provided below.

Frequency	Bedrock CALADPT	B&K DP0781
31.5 Hz	-0.3	0.0
63 Hz	-0.3	-0.3
125 Hz	-0.3	-0.3
250 Hz	-0.4	-0.4
500 Hz	-0.2	-0.2
1 kHz	0.0	0.0
2 kHz	-0.5	0.0
4 kHz	-1.2	+0.4
8 kHz	-6.5	-1.0
16 kHz	-5.5	-1.3

A3.2. Typical free field response for the BAMT2

The typical free field response of the BAMT2 microphone is specified in this section. Note that it is not feasible to perform actuator measurements with the BAMT2, as the microphone capsule is not detacheable. For instructions on how to perform periodic calibrations according to IEC-61672-3, which calls for a modified procedure for non-detacheable microphones, please refer to our manufacturer's instructions in Annex B.

The table below specifies the typical free-field response of the BAMT2 microphone. The typical free field response of the BAMT2 was measured by the manufacturer at 0 degrees incidence, based on average values of microphone capsules against a B&K 4180 response using a B&K 4228 Pistonphone.

BAMT2 Frequency	(1) Typical free field response
25 Hz	-0.45
31.5 Hz	-0.05
40 Hz	0.2
50 Hz	0.4
63 Hz	0.5
80 Hz	0.6
100 Hz	0.65
125 Hz	0.65
160 Hz	0.65
200 Hz	0.65
250 Hz	0.6
315 Hz	0.5
400 Hz	0.4
500 Hz	0.3
630 Hz	0.2
800 Hz	0.15
1 kHz	0
1.25 kHz	-0.2
1.6 kHz	-0.4
2 kHz	-0.6
2.5 kHz	-0.65
3.15 kHz	-0.6
4 kHz	-0.5
5 kHz	-0.45
6.3 kHz	-0.15
8 kHz	0.05
10 kHz	0.25
12.5 kHz	0.05
16 kHz	-0.7
20 kHz	-2.15

A3.3. Effect of the 32mm windscreen with the BAMT2

A 32mm-1/4" windscreen is supplied standard with each AM100 fitted with a BAMT2 microphone. The effect of this windscreen per octave band is given in the table below.

Frequency	Windscreen effect
31.5 Hz	0.0
40 Hz	0.0
50 Hz	0.0
63 Hz	0.0
80 Hz	0.0
100 Hz	0.0
125 Hz	0.0
160 Hz	0.0
200 Hz	0.0
250 Hz	0.0
315 Hz	0.0
400 Hz	0.0
500 Hz	0.0
630 Hz	0.0
800 Hz	0.0
1 kHz	0.0
1.25 kHz	0.1
1.6 kHz	0.1
2 kHz	0.1
2.5 kHz	0.2
3.15 kHz	0.3
4 kHz	0.5
5 kHz	0.7
6.3 kHz	0.8
8 kHz	0.7
10 kHz	0.4
12.5 kHz	0.3
16 kHz	0.2
20 kHz	-0.2

Annex B. Manufacturer's instructions for periodic testing according to IEC-61672-3.

B.1. Insertion of electrical signals for models with BAMT1 microphones

Periodic testing of the Bedrock AM100 in its capacity as a Sound Level Meter is to take place in accordance with the procedures as described in IEC-61672-3:2014. This standard requires the manufacturer to specify a means of inserting electrical signals.

For models featuring a BAMT1 microphone (BAMT1-MP201 or BAMT1-MP309), the procedure to enable insertion of electrical signals is to fit an adapter on the BAMT1 pre-amplifier (replacing the microphone capsule) with a capacitance that matches the MP201 and MP309 capsule (16 pF $\pm$ 2 pF). These are available from multiple vendors (contact Bedrock Audio for more information).

B.2. Insertion of electrical signals for models with BAMT2 microphones

For models featuring a BAMT2 microphone, a different procedure must be adopted, since the BAMT2 does not feature a detachable capsule. The BAMT2 is an electret microphone and preamplifier assembly; as a whole, this assembly is colloquially called "the micro-phone." The BAMT2 is removable from the device, allowing insertion of electrical signals into the system via the XLR connector. This enables test labs to carry out tests for which insertion of electrical signals is needed, in accordance with IEC-61672-3:2014.

For electrical insertion of signals into the XLR input, a test interface for the XLR input is available through Embedded Acoustics (article number 20179842). This interface matches the BAMT2 in signal level and impedance, and allows input of electrical test signals via a BNC connector.

B.3. Microphone response measurements

Microphone response measurements can be done using either of two methods:

- Using the the B&K 4226 Multi-frequency calibrator (set to pressure field calibration mode), taking free-field to pressure field corrections given in Annex A into account. Add these factors to the response measured with the mulittone calibrator to obtain the free-field response.
- Using an eletrostatic actuator (GRAS RA0015, GRAS 14AA or equivalent) taking ac-tuator to free-field corrections given in Annex A into account. Add these factors to the response measured with the actuator to obtain the free-field response.

B.4. Reference range and range adjustment

The reference range is the HI range. Once a range is selected through the "settings" module, no range adjustments take place during measurements and the AM100 acts as a single range instrument. Adjustment of the display range within the modules themselves does not have an impact on the measurement range, just on which range of values are displayed in graphs on the display.

B.5. Reference range for linearity measurements

The linearity of the AM100 is to be evaluated in the range 30 - 124 dB.

B.6. Wind spheres

The wind spheres are not part of the standard configuration and need not be considered for periodic testing. Wind sphere corrections are provided in Annex A for information purposes only.

B.7. Silicone diffuser ring

A silicone diffuser ring is provide to reduce the impact of reflections against the body of the instrument. The ring is mounted by placing it around the microphone preamplifier and pressing it against the instrument body. The diffuser ring must be mounted for any (acoustic) testing whenever the microphone and preamp are mounted on the device.

Annex C. Manufacturer's instructions for periodic validation of STI test functions

The STI test functions of the AM100 (STIPA and full STI) are based on algorithmic calculations of spectral and temporal signal patterns measured by the device's electro-acoustic front-end. The manufacturer has taken great care to ensure that the STI implementations are fully compliant with IEC-60268-16, and certifies compliance.

The nature of the STI algorithms is such that no degradation of the precision and accuracy of the measurent over time is to be expected, except when caused by one of the following factors:

- The frequency transfer function of the acoustic front-end (flatness) is modified and no longer compliant with class 2 requirements according to IEC-61672.
- The sensitivity of the acoustic front-end has changed so that the measured octave band leves are not longer sufficiently accurate. The A-weighted level must remain within 0.5 dB of its nominal value (level calibration).

Generally speaking, as long as the AM100 complies with requirements of IEC-61672, there is no concern regarding its compliance with 61628-16 (given the fact that the device type and algorithms have previously been shown to comply). For additional assurance a short series of tests can be done, as part of periodic check-ups of the device, to further ensure the continued validity and accuracy of the AM100 as a speech intelligibility meter. These tests are presented below.

Note that these tests are not suitable for verification of new implementations of the STI, in which case specific test vectors are needed to check proper implementation of the algorithms underlying the STI. The tests described here, in the form of checklists, are intended to periodically check STI meters for which the manufacturer has previously demonstrated compliance to IEC-60268-16.

In order to carry out the verification tests a Bedrock BTB65 TalkBox is needed, as well as a passive attenuator (or set of attenuators) that can be adjusted to 20, 40 and 60 dB of attenuation.

The aim of these tests is to verify that the electro-acoustic front end of the measuring system is within specifications. The instructions assume that the instrument under test is a Bedrock AM100 device.

Item	Goal	Check
Level accuracy	- Start RTA module. - Place microphone tip in tone calibrator (B&K 4231, Bedrock BAC1, Bedrock BAC2 or Cirrus CR:515.) - Select 1000 Hz band with the cursor in the RTA module. - Measure during 8 seconds. - The level at 1000 Hz must match the calibrator reference value (usually 94 dB) +/- 0.5 dB.	
Acoustic linearity	- Press +10 dB button on calibrator. Level in RTA module @ 1000 Hz must rise by 10 +/- 0.4 dB	
Flatness	- Remove the microphone and pre-amplifier, and connect the line-out of a Bedrock BTB65 Talkbox to XLR input into AM100, through attenuator set at 20 dB. The level into the XLR input of the AM100 will be approx. -32 dBU. - Select "pink noise" on talkbox. - In RTA mode, set blue channel to LZEQ - Check bands with cursor. All 1/3 octave bands in the range 100 Hz- 12.5 kHz must lie within + 3 dB of the value at 1000 Hz.	
Linearity	- Set attenuator at 40 and 60 dB (resp. -52 and -72 dBU). A-weighted level must reduce by the attenuation setting +/- 0.4 dB	
STIPA test	- Set attenuator at 40 dB (input level to the AM100 approx. -52 dBU). - Open the STIPA pro module. - Play that STIPA signal from talkbox (line out unmuted). - Run STIPA measurement until finished (25s) - STI must be at least 0.98 - Go to measurement details screen in the STIPA pro module. - column "Rel" must show only values between -2.0 dB and +1.0 dB.	