INSTRUCTION MANUAL

INSTRUMENT SOFWARE 4.0

Nor131 is a versatile IEC Class 1 sound level meter. The combination of only one large measurement range with a lot of parameters measured in parallel makes the sound level meter easy to operate and facilitate reliable measurements. The optional 1/1-octave and 1/3-octave analysis, statistics. level vs. time, reverberation time and STIPA calculations further enhance the range of application.

Nor132 is a IEC Class 2 instrument with similar features, except for a fixed microphone.

nor**131** nor**132**

nor**131**
nor**132**

Nor131/Nor132 User Guide – November 2018 Edition

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Finding the information you need

Thank you for choosing Norsonic! The sound level meters Nor131 and Nor132 have been designed to give you many years of safe, reliable operation.

The *User Guide* has been divided into fourteen chapters or sections. Each chapter provides different information. Depending on your requirements and your familiarity with sound measurements as such, you may find that you use some parts of this manual often and others not at all.

The very first chapter acquaints you with the Nor131/Nor132 and describes its features and possibilities. This may be a good starting point so that you know more about what to look for and what you maybe should learn more about.

The next section provides a closer look at the instrument with a presentation of all major parts and the keys of the keyboard.

Calibration is a vital point ensuring that your measurements are sufficiently correct for the purpose. Therefore, a separate chapter has been devoted to this.

How to measure with the sound level meter is described in the following chapters. The first measurement description outlines the use of the instrument as a simple sound level meter. The second description extends the description to also include frequency analysis.

Detailed information about the instrument is found in the chapter covering Technical specifications.

Note that the instruction manual describes a fully equipped instrument. Your version may not have all the optional extensions available. Extensions may, however, be installed as retrofit any time.

Our objective with this manual has been to address your goals and needs. Please let us know how well we succeeded!

Contents

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Introducing the sound level meters Nor131 and Nor132

The sound level meter Nor131 and Nor132 comes in different versions dependent on the number of optional functions installed. Even in the basic version the instrument is able to measure an extensive number of functions characterizing the measured sound. This include:

- SPL The Instantaneous Sound Pressure Level
- L_{MAX} The Maximum Sound Pressure Level
- L_{MIN} The Minimum Sound Pressure Level
- L_{eq} The Integrated Equivalent SPL
- L_{E} The Sound Exposure Level
- L_{PEAK} The Maximum Peak Level

The time-weighting can be F, S or I. If your instrument is equipped with parallel time constants (option 7), the list of functions measured simultaneously can be expanded to include functions with F, S, and I time constant at the same time. While the measurement is running the instantaneous SPL is available for inspection, but as soon as the measurement is terminated, the SPL becomes meaningless and therefore not listed in the result tables.

Two spectral weighting functions: A- and C- or A- and Z-weighting, are simultaneously available for all functions listed above including the L_{PEAK}.
The Z-weighting defined in the International Standard IEC 61672-1 replaces Lin or Flat as these have not been properly defined by any standard.

Time and date. The instrument contains a real time calendar and clock. For each measurement, the time and date for the measurement as well as the measurement duration, are stored with the results. The measurement duration may be set from 1 second to 200 hours.

No range setting. Only one level range with a dynamic range in excess of 120 dB makes the setup easy and ensures reliable measurements in all situations – especially useful for unattended measurements. With the normal microphone this means a measurement range covering levels from the self-noise of the microphone (typically less than 20 dB A-weighted) to sound with peak values up to 140 dB.

Storing and retrieving of results. The results from a measurement may be stored – either automatically or manually – in the non-volatile memory of the instrument together with information of instrument set-up. The information may later be displayed on the instrument screen or transferred to an external device like a PC.

Back-erase. The back-erase feature deletes the ten most recent seconds of acquired data prior to pressing the pause button. The measurement may be resumed after a second operation of the button.

Microphone included. Both instruments are delivered fully equipped with a microphone. Nor132 comes with a fixed microphone and Nor131 with a detachable microphone. This microphone has a preamplifier of the IEPE-type which allows an extension cable (not included) up to 5 m between the preamplifier and the instrument.

Battery operation. The instrument is normally powered from internal batteries which typically last for more than eight hours. For noise monitoring and other long time operation the instrument may be powered from an external DC-supply or mains adaptor Nor340.

Setting up is easy to do. After you've defined the duration of the measurement, all you need to do before you press the start key, is to define a few parameters like the time constant and whether to use Z- or C-weighting as the secondary spectral weighting function. The selection last used is automatically selected when you power the instrument, but you may also store and later retrieve a particular setup. During the measurement you have instant access to the analysis.

Excellent for noise monitoring. With the Nor131 and Nor132 you are able to do nearly all types of noise measurements – community noise, industrial hygiene, product control, noise mapping and more.

But, you don't need to attend the measurement sessions all the time. In fact, the instruments are able to do a lot on their own.

The measurement time can be pre-set to e.g. 30 minutes or an hour and the instrument put in a mode where it measures and stores the results, then starts measuring again, stores the results, starts measuring again and so on. Each measurement will be stored in a separate file, but all files are stored in the same directory, which has the name of today's date. In this way the instrument will measure the periods you need.

The instrument will spend a little time storing the data. Therefore, if you start the session with hourly measurements exactly on the hour, long "store and go" sessions will – after a while – experience a small, but significant time shift, so that each period no longer starts exactly on the hour. The synchro mode solves this problem (standard in all models). When selected, this mode stops the measurement slightly earlier to provide time for store and restart. By sacrificing a little at the end of a measurement the synchronisation with the time of day is retained, a feature important to many of our customers (E.g L_{DEN}).

Accessories. The Norsonic product range contains a wide range of equipment and accessories for use with acoustic measurements and noise monitoring. We supply enclosures for permanent monitoring installations, environmental cases for semi-permanent installations, microphones for applications in tough environments, cables, modems, weather stations and post-processing software. A detailed presentation of this is available in a separate leaflet and on the Norsonic home page: www.norsonic.com.

Real time frequency analysis

As an option, the instrument may be delivered with octave or third-octave band, real time frequency analysis. The analysis is done for all filter bands in parallel and centre frequency span the range 8 Hz to 16 kHz in octave bands or 6.3 - 20 kHz in third-octaves.

The same functions as for the frequency weightings – except peak – are available for each octave band:

- SPL The Instantaneous Sound Pressure Level
- L_{MAX} The Maximum Sound Pressure Level
- L_{MIN} The Minimum Sound Pressure Level
- L_{eq} The Integrated Equivalent SPL
- L_E The Sound Exposure Level

The graphical level indications for all frequency bands measured are visible in one display with no need for horizontal scrolling.

Furthermore, the instrument measures the full frequency range – every time! Given the high dynamic range (120 dB), which eliminates the need for adjusting the gain, there is very little to set up before a frequency analysis can be made – successfully!

The measured functions are also available in tabulated form as numerical values. By pressing the **TBL** key during or after a measurement you have instant access to the numerical version. Another push on the key will restore the graphical display.

Level statistics

The optional extension *level statistics* adds statistical functions to the instruments. The F-time weighted level is sampled ten times per second and the statistical distribution of these samples are calculated for each frequency weighting and each octave-band, if available. The F-time weighting is used irrespective of which time weighting the instrument otherwise employs.

The class width is always 0.2 dB to ensure sufficient resolution and the results are presented in the form of eight percentiles. One of these percentiles is subject to user-definition and can be set to anything from 0.1% to 99.9%, both extremes included. You don’t have to define the percentile prior to the measurement. You may redefine the percentile as many times as you like – even after the measurement – for every frequency band measured!

However, for results stored in the instrument’s memory, only the selected percentiles will be available to keep the amount of stored data lower.

The options available.

- Opt. 0: LTmax5 and LeqI measurements according to German Standards.
- Opt. 1: 1/1-octave real-time filters 8–16.000 Hz
- Opt. 2: Statistical calculation
- Opt. 3: Level vs Time measurements
- Opt. 4: 1/3-octave real-time filters
- Opt. 5: Reverberation calculation
- Opt. 6: STIPA calculations
- Opt. 7 Parallel F, S, I time constants

The fixed percentiles. The fixed percentile levels offered by the statistic option are 1.0%, 5.0%, 10.0%, 50%, 90%, 95% and 99%.

For the statistical sampling the instrument makes use of the F time weighting.

Note! The codes are unique for each instrument and will not work in other instruments. Do not change these codes as you will then loose the optional extensions installed and activated!

Reverberation Time

The optional extension 5 for the Nor131/132 permits measurements of the reverberation time based on impulse excitation

Typical applications include the measurement of reverberation time as required for eg. classrooms and kindergardens

The reverberation time is measured simultaneously in every frequency band in the range 63 Hz to 8 kHz when 1/1-octave bandwidth is selected, and in the range 50 Hz to 10 kHz for 1/3-octaves.

Speech Intelligibilty

It is now possible to use the Nor131 and Nor132 meters for analysing the Speech Transmission quality in public areas using the STIPA method following the latest revision of the standard IEC 60268-16:2011.

The method can be used to compare the speech transmission quality at various positions and under various conditions within the same listening space.

A measurement in one listening position takes about 13 sec. Unlike many other STIPA measurement systems, the implementation with the Nor131/132 can also correct the results for the background noise.

In addition all calculated indexes are displayed, not only the single STIPA value. This feature is valuable for engineers optimizing the room acoustics in public spaces or other areas where the speech quality is important.

The principle of optional extensions

The capabilities and setup options of your sound level meter will depend on which of the available extensions it has been equipped with.

Extensions are modules – made as software, in the instrument or e.g. as software for your PC – available for your instrument. Norsonic extensions are always optional and hence often referred to as options. In this way you do not have to pay for features you're not going to use anyway.

However, you may find that your tasks are expanding into new areas of acoustics as time goes by. Therefore a typical Norsonic extension will be available for installation as retrofit.

Check which extensions are installed

Unless you are certain about the extensions installed in your sound level meter, we recommend that you spend a little time looking into the matter.

The extension menu. Press **SETUP** > **1** > **0**, although you won't find the 0 listed as an option in the Instrument setup menu.

The menu contains a unique Id code which identifies the very individual instrument. In addition, the menu contains three codes. These codes enable the extensions activated for this instrument. The codes take the Id number into account and are valid for this individual instrument only.

- To leave the menu without restarting, press exit and then **ENTER** twice.
- To leave the menu and let changes made take effects, move the cursor to "set codes" and then press enter. Note that the instrument will restart as a consequence of this.

Taking a closer look at the instrument

You may have to assemble the instrument the first time you use it. Be sure to take utmost care when mounting the microphone cartridge onto the preamplifier.

Always keep the preamplifier disconnected whenever you are screwing the cartridge onto the preamplifier and screw only finger tight! We recommend that the microphone is always mounted on the preamplifier as this will prevent dust and dirt to enter the insulator around the sensitive signal terminal on the microphone.

The instrument is powered from four AA size batteries which are inserted as shown on the figure.

On the use of batteries

The sound level meter comes with four AA batteries (LR6, 1.5V each). Battery lifetime is typically 8–12 hours (depends on measurement mode and brand of batteries). If you switch to lithium batteries the life time will increase to 15–20 hours. The use of alkaline or lithium batteries is strongly recommended to avoid leakage.

Rechargeable batteries may also be used, but with reduced operating time. Connecting an external DC-source (11–16V) to the instrument will not charge rechargeable batteries, but power the instrument in lieu of the internal batteries.

If the instrument is in regular use, always keep the batteries in the instrument. Even nearly flat batteries will contain sufficient power to supply the internal calendar/clock. During change of batteries the clock is powered by an internal capacitor. The capacitor will supply the clock for nearly an hour, but we recommend installing new batteries immediately after the old batteries were removed.

If the instrument is stored for a prolonged period of time, we recommend removing the batteries to avoid damage from leaky batteries. However, you then need to adjust the calendar/clock before you start to use the instrument again. Note that the date is used as a part of the automatically generated file name when measurement results are stored in the instrument. Having set a wrong day may therefore lead to mixing old and new measurements.

Data are stored in a non-volatile memory and will retain its content independent of the batteries.

Switch on the instrument

Press the **ON**/**OFF** button in the lower right corner of the instrument. A bargraph is displayed during initiation of the instrument. A second operation will switch the instrument off.

Battery Voltage vs. Time

The sound level meter offers a graphic presentation of the battery voltage-versus-time history.

To display the battery voltage vs. time:

- Press the **BATT** key. Press again or **ENTER** to exit the menu.

The display will now indicate the combined voltage of the four batteries and the use-time elapsed. Each pixel corresponds to seven minutes in the horizontal direction and 0.25V in the vertical direction.

When the combined battery voltage drops below 3.9 V, a battery low indicator appears in the display and the instrument will start to shut itself off. Any ongoing measurement will be terminated and the results stored in a directory called BATLOW. Memory contents is retained without the use of electrical power (flash memory). Upon installing fresh batteries, the instrument will at start-up ask the operator for the confirmation to store the previous measurement in the normal measurement directory.

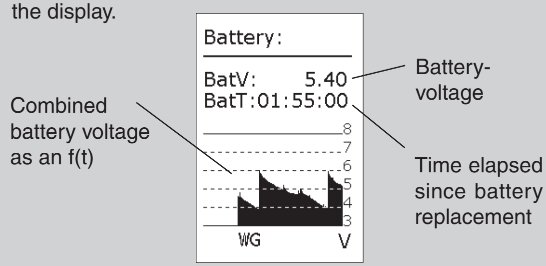
If the instrument is connected to an external DC-source, the battery voltage vs. time diagram will be halted and information about the external supply (Ev) is given.

If powered from internal batteries and left unattended and unoperated, the sound level meter will switch itself off after ten minutes. A warning will be displayed on screen during the last minute before switch-off. However, this does not apply if the instrument is measuring (including being paused during a measurement), or when powered from an external source.

See the “Technical specifications” on page 69 for details.

Battery voltage

Battery voltage is presented graphically as f(t). Press the batt key to produce this display and press again to exit the display.



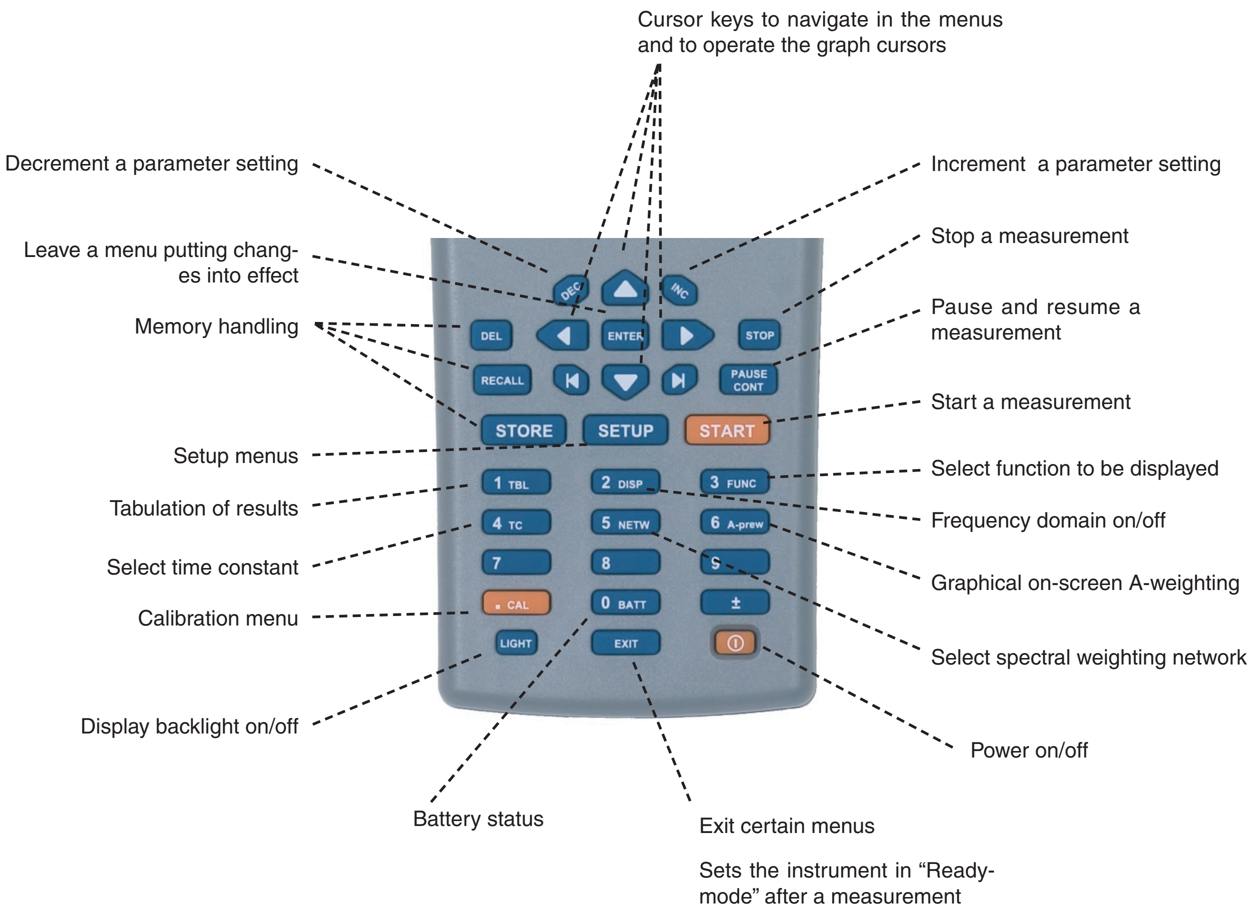
No recharging. Connecting an external DC-source (11–16V) to the instrument, will not charge rechargeable batteries, but power the instrument in lieu of the internal batteries

Front panel overview

Navigating in the menus. Observe the following general guidelines applicable to every instrument menu:

- To navigate between editable parameter fields in the menu, use the cursor keys
- The editable field currently selected is shown inverted (white text on black background)
- Use the Modifier keys below the display (the **INC** and **DEC** keys) to increment or decrement the current setting of the parameter. Alternatively use the keypad to key in the required value, whenever applicable. The # sign will appear in the lower line of the display whenever the instrument accepts numerical inputs
- If you use the numerical keypad, be sure to press enter (Request for Enter is marked as #E in the display) before moving to the next field to alter. This is not needed when you use the **INC** and **DEC** keys.
- To leave a menu putting changes into effect, press enter.
- There is no cancel function available.

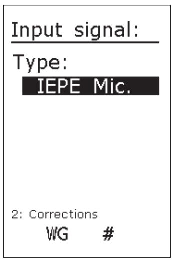
The front panel keys



Powering the microphone preamplifier

The detachable microphone preamplifier on Nor131 is powered through the terminal on the TNC-connector. If the input terminal is connected to other types of signal sources, the supplied current may be switched on or off dependent on the application. To set the supply current on to accommodate the normal microphone with preamplifier, do as follows:

- Press **SETUP** > **1** (Instr.) > **4** (Input) and use the **INC** or **DEC** to select Mic.
- Press **ENTER** to leave the menu putting changes into effect.



When the microphone is selected, a compensation for the signal attenuation in the microphone preamplifier is automatically activated. See the section about *Calibration* for further information.

- Press **SETUP** > **1** (Instr.) > **4** (Input) and use the **INC** or **DEC** to select LINE. Press **ENTER** to leave the menu putting changes into effect.

The microphone is prepolarised and needs no externally supplied polarisation voltage.

As described the default sensor is:

- IEPE mic. – In this mode the instrument supply a constant current of 3mA on pin 4 on the Lemo microphone socket.

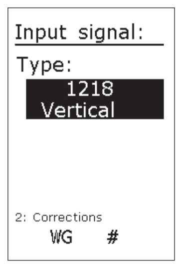
  This is used when connecting IEPE accelerometers or preamplifiers with pre-polarized microphones.

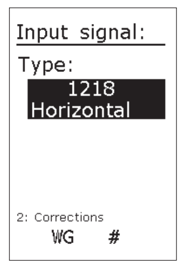
  Use this setting for the Nor1228 (1229) (standard delivery).

  All preamplifier corrections and other corrections like windscreen, random field etc. are turned off.

It is also possible to use the outdoor protection kit:

- Nor1218 Vertical – vertical direction. This mode is used when connecting Nor1218 with the sound propagating in the vertical direction, typical used when measuring air craft noise.
- Nor1218 – Horizontal direction. This mode is used when connecting Nor1218 with the sound propagating in the horizontal direction. Typically used when measuring community noise, like traffic or industry noise.





In addition there is support for other sensors:

- Line – This mode is used for connecting an electrical signal directly to the instrument.

  The IEPE current is turned off in this setting

  All preamplifier corrections and other corrections like windscreen, random field etc. are turned off.

Setting the time and date

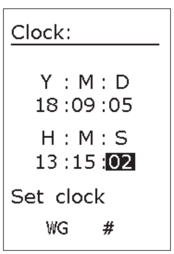
To set the time and date:

- Press **SETUP** > **1** (Instr.) >**2** (Clock). Use the cursor keys to navigate in the menu and **INC** and **DEC** to alter a setting or use the numerical keypad to key in a value. Numerical inputs must be terminated by enter to enable navigation between the parameter fields again. Press **ENTER** to leave the menu putting changes into effect (i.e. setting the time and date).

The following letters are used for day and time:

Y : M : D = year : month : day
H : M : S = hour : minute : second

Note that only two digits are used to designate the year.



Calibrating the instrument

Calibration is the normal way of ensuring that the sound level meter measures the level with sufficient accuracy. To calibrate we need a sound calibrator.

The use of sound calibrators dates back to the days when it was easier to design a stable sound calibrator than a stable sound level meter. Today, sound measuring instruments are, in general, as stable as the sound calibrators.

However, measuring microphones are very delicate devices designed to fulfil very rigid specifications. This makes them vulnerable and subject to damage unless proper care is taken.

One may therefore say that a sound calibrator is just as much a verification of proper operation as it is a device of adjusting the sensitivity of sound measuring instruments.

The Nor131 and Nor132 are calibrated by means of menus and key pushes – there is no need for a screwdriver to turn a potentiometer!

When to calibrate

Calibration of the sound level meter should preferably take place before and after a measurement session is commenced, or whenever required by applicable standards. If you know the combined sensitivity of the microphone cartridge and the preamplifier, you may key this in using the numerical keypad. However, doing so will never replace calibration with a sound calibrator, as the sensitivity adjustment procedure will be unable to reveal possible microphone, preamplifier or extension cable malfunctions.

No need to adjust the full scale setting

Since the sound level meter has a dynamic range of more than 120 dB, the 80 dB bar graph range is a display limitation only. Hence, you won't have to bother with setting the full scale before you enter the *Calibration* menu.

Furthermore, since the sound level meter automatically enters C-weighted mode, you won't have to bother with the calibrator frequency either if your calibrator apply a frequency between 250 Hz and 1 kHz.

However, you may have to adjust the display top scale setting to ***see*** the top of the bar graph. Use the **INC** and **DEC** keys for this after the calibrator is switched on and ***before*** you enter the calibration menu.

The microphone and the preamplifier should be considered as one unit. The preamplifier attenuates the open-circuit voltage from the microphone in order allow sound with peak level up to 140 dB to be measured without overloading the preamplifier of the IEPE-type. The attenuation in the preamplifier is typically 6 dB. This attenuation is specified in a correction menu. The calibration value should therefore be close to the open-circuit sensitivity for the microphone.

Carrying out the calibration

You will need a sound calibrator of sufficient accuracy, i.e. a class 1 or class 2 sound calibrator as defined by the International standard for sound calibrators: IEC 60942. In general we recommend a class 1 calibrator for Nor131 (such as the Norsonic sound calibrator Nor1251 or Nor1253) and a class 1 or class 2 calibrator (Nor1252) for Nor132. Do as follows:

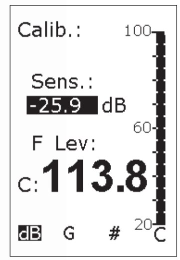
1 **Mount the calibrator.** Mount the sound calibrator onto the microphone as shown to the right. Switch on the sound calibrator and wait until the level has stabilised. Information on how long time this will take should be available from the documentation accompanying your sound calibrator. Adjust the displayed range by the **INC** or **DEC** keys to display the level (The numeric value is independent of this adjustment).

2 **Enter calibration mode**. Press the **CAL** key to gain access to the Calibration menu. The display will typically look as shown on the figure.

3 **Know the output level of your sound calibrator.** Some sound calibrators have an output level of 94dB, while others (like the Nor1255 which is used in the example to the right) have an output level of 114dB. Unless you know the output level of your sound calibrator you won't be able to know what level the measuring instrument is supposed to show. The output level is normally printed on the sound calibrator or stated in its accompanying user documentation or calibration certificate.

The sensitivity specified in the Calibration menu is the microphone sensitivity in dB relative to 1 volt/pascal, e.g. 50 mV/Pa corresponds to –26.0 dB.

Use **INC** / **DEC** keys to adjust the sensitivity or use the numerical keypad to key in the sensitivity



Field calibration. The recommended sound calibrator for verification of the sound level meter Nor131 is the Norsonic Nor1255, class 1 calibrator, with a nominal sound pressure level of 114.0 dB @ 1 kHz.
The recommended sound calibrator for the sound level meter Nor132 is also the Norsonic Nor1255.

In order to compensate for diffraction effects around the microphone, we recommend adjusting the sound level meter to indicate 113.8 dB (diffuse/random correction off).

If other types of calibrators are to be used for the calibration, we recommend adjusting the sound level meter to indicate the following levels referred to the sound pressure level acting on the microphone's diaphragm (random incidence correction off):

f [Hz]	125	250	1000	4000	8000
Corr. [dB]	0.0	0.0	–0.2	-0.8	–2.8

If the random incidence (diffuse) correction is on, use the sound pressure level stated on the calibrator for any of the above mentioned frequencies.

The correction is activated and deactivated in the Corrections menu. Press **SETUP** > **1** (instr.) > **4** (Input) > **2** (Corrections). Navigate in the menu using the arrow keys and use **INC** or **DEC** to activate/deactivate the Random setting. Activated Random setting is indicated by an R in the lower line of the display.

4 Free-field microphones require lower settings. Be aware of the fact that instruments using free-field microphones – as normally delivered with Nor131 and Nor132 – shall be adjusted to a value slightly lower than the output level of the sound calibrator. For a half-inch cartridge this will typically amount to 0.2 dB lower for calibrators producing a 1000 Hz calibration signal (e.g. the sound level meter should then be set to 113.8 dB when using a 114 dB @ 1000 Hz sound calibrator) Other frequencies will require different correction values, see the Field calibration side bar for more on this.

5 Set the sensitivity. To set the sensitivity correctly use the **INC** / **DEC** keys (below the display) while at the same time watching the level read-out. Alternatively, you may key in the required sensitivity using the numerical keypad. Always use a figure before the decimal sign (e.g. 0.3). Once the correct level reading is established press **ENTER** to leave the menu.

The windscreen correction **W** is automatically switched off during calibration.

For line input selected, by setting the sensitivity to -26.0, a reading of 0 dB corresponds to 1 microvolt and 120 dB to 1 volt on the input terminal.

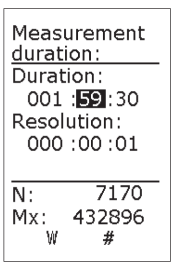
Sound measurements

Due to the wide measurement range and all the functions measured in parallel, the Nor131 and Nor132 are easy to use. The only thing you really need to set up is the measurement duration, which at least must be set up to match the amount of time you intend to be measuring. If it is set to a longer time, this will constitute no problem – just press the stop key when you want to terminate an ongoing measurement.

However, you should consider the settings of the time constant and the spectral weighting (C- or Z-weighting, see *Setting C or Z as spectral weighting network* for more on this) also, but once they are set, the instrument will remember these until they are changed to something else.

Setting the measurement duration

Press **SETUP** > **2**. Use the cursor keys to move the cursor to the requested field. The fields are the time in hours: minutes: seconds. Adjust the value by using the **INC**/**DEC** buttons. Alternatively, key in the numeric value and press **ENTER**. To leave the menu press the **ENTER** key again.



Navigating in the menus. Observe the following general guidelines applicable to every Nor131/Nor132 menu:

- To navigate between editable parameter fields in the menu, use the **CURSOR** keys
- The editable field currently selected is shown inverted (white text on black background)
- Use the **ARROW** keys to right of the display (the **INC** and **DEC** keys) to increment or decrement the current setting of the parameter. Alternatively use the keypad to key in the required value, whenever applicable. The # sign will appear in the lower line of the display whenever the instrument accepts numerical inputs
- If you use the numerical keypad, be sure to press **ENTER** before moving to the next field to alter. This is not needed when you use the **INC** and **DEC** keys.
- To leave the menu putting changes into effect press **ENTER**.
- There is no **CANCEL** function available.

Setting the time constant

If your instrument is *not* equipped with the optional extension 7 – parallel time constants, you have to specify the time constant to be used in the measurement. The time constant is used for the SPL, the L_{eq}, the L_{E} , the L_{MAX} and the L_{MIN} measurements, but not for the L_{PEAK}. The exceedance level, L_{n},, always apply the F time weighting and is independent of the setting.

To set the time constant press the **TC** key until the required time constant appears in the display. To see this, be sure to operate the **FUNC** key until any of the functions SPL, L_{eq}, L_{E}, L_{MAX} or L_{MIN} appears in the display first.

If your unit is equipped with multiple time constants, you need not bother with this.

Instruments with multiple time constants

Instruments equipped with multiple time constants will always employ all three time constants (F, S and I) for all measurements. The time constants apply to the SPL, the L_{MAX} and the L_{MIN} functions. L_{eq}, L_{PEAK} and L_{E} do not make use of time constants. Units configured for German-speaking markets will also measure the L_{eqI}.

The presence of multiple time constants eliminates the need for setup of the time constant.

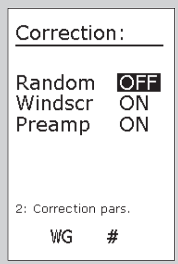
Corrections In order to improve the accuracy, you may add corrections to the frequency response of the sound level meter. Press **SETUP** > **1** > **4** > **2** (corrections) to access the menu for the corrections.

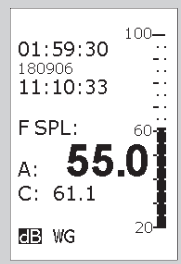
Random response

The microphone normally supplied with the sound level meter is made optimal for sound approaching the microphone from the front (Flat free-field response). This will reduce the sensitivity for sound in other directions. In order to obtain a flat random-response, a spectral correction may be applied. Marked with an **R** in the display when selected.

Windscreen correction

Wind noise may be reduced by mounting the wind-screen Nor1451 supplied with the instrument. We recommend to switching on the windscreen spectral correction while the windscreen is mounted. Marked with an **W** in the display when selected



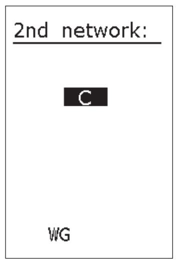


C or Z as spectral weighting

The Nor131 and Nor132 have three spectral weighting functions in addition to the optional filter bands. These are A-, C- and Z-weighting. A-weighting is always selected, the second network has to be selected C or Z. The Z-weighting is a replacement for the previous Flat or Linear spectral weighting functions.

To specify whether to use Z- or C-weighting:

- Press **SETUP** > **1** (instrument) > **3** (2nd netw) and navigate in the menu as usual. To leave the menu, press the **ENTER** key.



Making a measurement

To start a measurement:

- Press the start key. The **R** in the display indicates that a measurement is running.

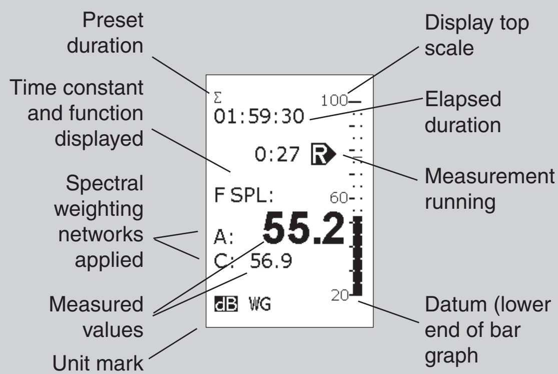
To temporarily halt an ongoing measurement:

- Press the **PAUSE/CONT** key.

To resume a paused measurement:

- Press the **PAUSE/CONT** key again. Upon resuming, the instrument will go on measuring until the total measurement time elapsed equals the preset duration. Observe that data acquired the ten seconds immediately preceding the pause will be erased because of the back-erase function (see below).

The sound level meter display



To terminate an ongoing measurement:

- Press the **STOP** key.

To resume a terminated measurement:

- To resume a terminated measurement press the **PAUSE/CONT** key again. Upon resuming the instrument will go on measuring until the total measurement time elapsed equals the preset duration. When a terminated measurement is resumed, the back-erase feature (see below) will not be activated.

To display other functions measured,

- Use the **FUNC** key. For the German-speaking markets these functions will include L_{eqI} and T_{Max5}

To adjust the display top scale:

- If the bar graph fails to match the level measured use the **INC** and **DEC** keys to alter the display top scale setting. (Note: This will also affect AC-out level, see *chapter 8, Signal out.*)

To switch between the spectral weighting functions:

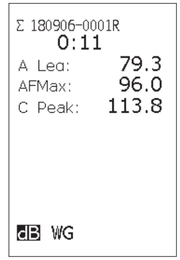
- Use the **NETW** key to switch between A-weighted and C- or Z-weighted or the A-weighted and the C–A (Z–A) weighted functions.

To produce the results in tabulated form:

- Press **TBL** to produce a result table. See *Displaying the result tables for more on this.*

User defined table

The user may compose a user defined numeric table consisting of just the needed parameters for a given application. A typical task is a table consisting of only L_{Aeq} and Peak C. The parameters is selected in **SETUP** > **1** (Instrument menu) > **8** (Misc. parameters). Use **INC/DEC** key to select parameters on/off. Use arrow keys to move to the desired field. Use **NETW** key to move between A, C (or Z) and Ln% table values. The table is accessible using the **TBL** button if you are in the graphical Sound Level Meter display (Not L/t or L/f display). Please note that all the other standard tables accessible from the graphical sound level meter display are replaced by the user defined table. If you need to revert to the standard tables, just enter the user defined table setup and turn off all parameters.



Resuming an ended measurement

Assume that you have set up the instrument to measure for 5 minutes and that you start the measurement. After 5 minutes the measurement will end since the measurement time elapsed equals the preset duration. The measurement has now ended successfully, as opposed to if you press the stop key to forcefully terminate an ongoing measurement.

If you now press the **PAUSE/CONT** key, the instrument will resume the measurement and go on measuring for another 5 minutes so that the total measurement time assumes 10 minutes, i.e. twice the initial setting. If you do this again, the total measurement time will be 15 minutes, i.e. three times the initial setting and so on.

This way of prolonging a measurement will not activate the back-erase feature (see below for more on this).

The back-erase feature

When you press the **PAUSE/CONT** key during an ongoing measurement, the instrument will temporarily halt the measurement. Pressing the key again will cause the instrument to resume the measurement while at the same time erasing the data acquired during the last 10 seconds immediately preceding the pause. This feature allows you to remove untypical events from the measurement.

Make a level average!
The resume function may be used to obtain the averaged level at different locations or conditions. As an example, you want to obtain the averaged value from three measurement positions. Make the first measurement by pressing START. Start the second and the third measurement by pressing the **PAUSE**/**CONT** key. The L_{eq}-value after the last measurement will be the averaged level.

If the measurement has been running for less than 10 seconds when you press the **PAUSE/CONT** key, the entire measurement will be erased upon resuming the measurement.

If less than 10 seconds have elapsed since the last time you resumed a paused measurement, only the part of the measurement acquired since the last resume will be erased. Data acquired earlier are assumed to be accepted by you.

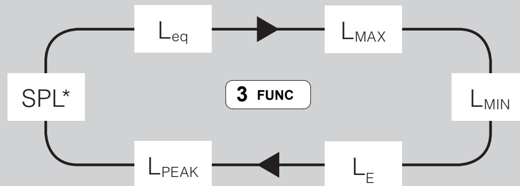
The measurement time elapsed counter will be updated to reflect the back-erase. Note that the statistics buffers (optional extension) will be updated similarly.

Displaying the functions measured

The instrument measures the SPL, L_{MAX}, L_{MIN}, L_{eq}, L_E and the L_{PEAK}. Note that the SPL, L_{MAX} and L_{MIN} are all measured with the selected time constant while the rest do not make use of the time constant at all.

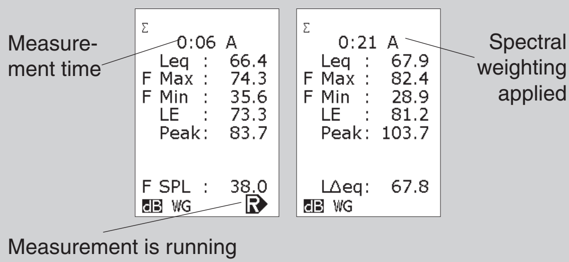
During measurements the numeric SPL value is updated every second. Once the measurement is over, the SPL becomes meaningless. A single SPL value cannot be used to characterise the measurement unless it represents some kind of maximum, minimum or time-integrated average. It is thus not accessible post measurement.

The functions are available sequentially
To display a certain function, press the **FUNC** key repeatedly until the function appears. The sequence is as follows:



*Accessible during measurement only – not after!

The result table
Once you've pressed the **TBL** key, the display will typically look like this:

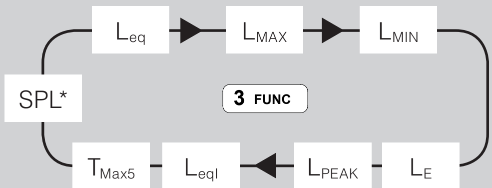


Observe that once the measurement no longer is running, no SPL value is available.

Units *not* configured for the German speaking markets will have tables not containing the I L_{eq} (L_{eqI}) and T_{mx5} values.

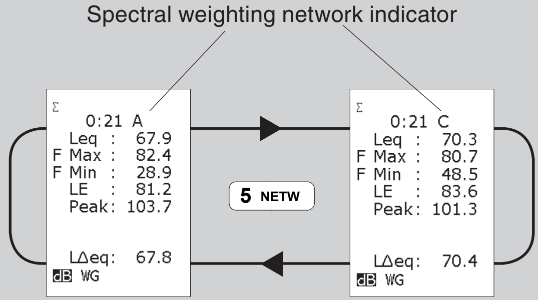
Units configured for the German-speaking markets

If also equipped with LeqI and TMax5, the sequence is as follows (press **FUNC** key repeatedly):



*Accessible during measurement only – not after!

Displaying the result table using the other spectral weighting function



To return to READY mode, i.e. how the instrument behaved before the measurement was started, press the **EXIT** key. You will be prompted to store the data or press **EXIT** again. In both cases the instrument will go back to READY mode. The SPL will now be displayed again.

If you choose not to store the data (i.e. you did press **EXIT** a second time) the measured data will be lost beyond retrieval. Storing is dealt with later.

Displaying the result tables

As an alternative to the above procedures, you may display all the data in a single table. Press the **TBL** key to produce the result table. This feature is available during as well as after a measurement. Do not forget that SPL values are only shown during a measurement – never once the measurement is over!

To produce the table of measured results using the other spectral weighting function press the **NETW** key while in the table.

There are two spectral weighting functions available A- and C- or A- and Z-weighting. The latter should be set by you prior to the measurement.

Even before you enter the table you may use the **NETW** key to view the results of applying the two spectral weighting functions.

The actual spectral weighting function used in the table depends on the setting active before the **TBL** key was pressed. Press **NETW** to toggle between primary and secondary weighting function.

Statistics – displaying the percentiles

Instruments equipped with the optional extension 2 – statistics – will measure the statistics every time. This cannot be switched off.

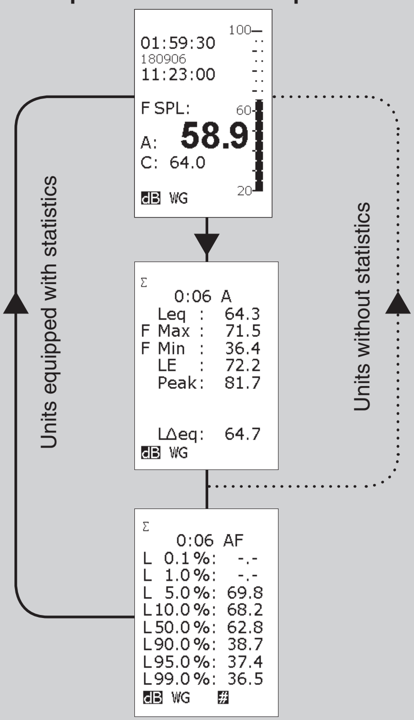
The sampling for the statistical calculations is made with F time constant and the class width is 0.2 dB over the entire 120 dB dynamic range – always!

You may think that storing all these data will require a huge memory, and you're absolutely right. Therefore, we refrain from that, we store just eight percentiles instead. Seven of them are fixed and one is user-editable.

Your user-editable percentile can be set to anything in the range 0.1–99.9 %, both extremes included. The procedure is explained overleaf.

Only measured data, but not data retrieved from stored values may be subject to changes in the user-defined percentile. This means that as long as you have the last measured data in the display, you may change the percentile as many times as you want, but once the data are stored and later retrieved any changing of the user-editable percentile is no longer possible.

The percentiles table sequence

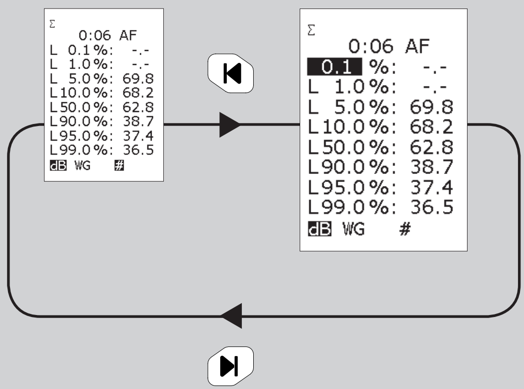


Editing the user-defined percentile

To enable the editing the instrument must display the percentiles table. In the percentiles table press the ⏮ key (the end-left key) to enable this.

Once the percentile field is shown inverted (with white text on black background) the text can be edited using the numerical keypad or **INC** and **DEC**. If you use the numerical keypad you must press **ENTER** to confirm that you have keyed in the new percentile.

To terminate the editing process press the ⏭ (end-right key). The corresponding percentile value will now be displayed, given that the measurement duration was long enough to provide a sufficient number of samples. Using **ENTER** will not work here.



The user-defined setting applies to both spectral weighting settings. The two cannot have individual settings!

Displaying the percentiles table

To produce the percentiles table press **TBL** to enter the result table and **TBL** again to enter the percentiles table. Units not equipped with the statistical extension will exit the table upon the second push on **TBL** and return to the sound level meter display mode. The sequence is shown in the side bar.

Storing the acquired data

The Nor131/Nor132 has a large, non-volatile memory to hold the measurements. The memory structure resembles the memory structure of a personal computer in the sense that both use folders and files.

All aspects of the memory handling are discussed in detail in *Memory handling.*

Missing percentiles? There may be percentiles that fail to produce values in the table. This is because you have not measured for a time long enough to provide the necessary number of samples.

Statistical sampling – which should not be confused with the sampling of the analogue-to-digital conversion – takes place 10 times a second. For comparison, the sampling of the analogue-to-digital conversion runs at 48 kHz!

Since the statistical sampling takes place 10 times a second, it will take 10 seconds to produce 100 samples. You will need at least 100 samples to be able to calculate the 1% percentile. Likewise, for the 0.1% percentile the minimum time required will be 100 seconds.

Frequency analysis

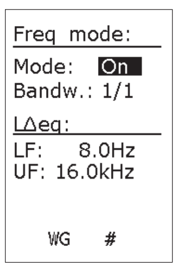
As an optional extension 1, you may have your Nor131 or Nor132 equipped with parallel octave band filters. By adding the optional extension 4, third-octave band filters become available as well. The frequency range, expressed as centre frequencies, is 8–16 000 Hz for the octave band filters and 6.3–20 000 Hz for the third-octave band filters.

When you make a frequency analysis, this comes in addition to the traditional sound level measurements as described in the chapter *Sound measurements*.

Setting up

The frequency range is fixed and cannot be changed by the user. Bearing in mind that the dynamic range of the instrument is in excess of 120 dB and that the instrument measures a fixed set of functions (which cannot be altered by you), there is not much left to set up before the instrument is ready to make a frequency analysis, besides selecting the band width. This is selected by pushing **SETUP > 3 :** Frequency.

In the Frequency mode menu you can select between 1/1- or 1/3-octave band width and switch the filters ON/OFF.



Statistics

If your instrument is equipped with the optional extension *Statistics*, the statistical functions will be calculated for each octave or third-octave band, as for the ordinary frequency weightings A- and C- or A- and Z-weighting.

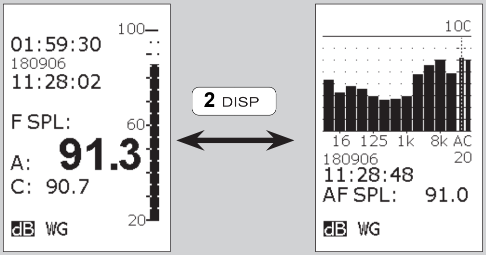
Displaying the spectrum

Press the key **DISP** to display the octave-band spectrum. This key is used to toggle between the normal display and the graphical octave-band display.

Since no measurement has been made the only function producing frequency band bar graphs will be the SPL with the selected time constant. If you press the **FUNC** key this will produce empty displays only. If you did, just keep pressing the **FUNC** key until the SPL reappears.

> When the frequency analysis extension has been installed, frequency analysis will be made during every measurement. This means that the instrument will combine a “traditional” sound level meter measurement and a real time frequency analysis in octave or third-octave bands. Whether the instrument is set to show the spectrum or the classic sound level meter display will not affect the measurement.

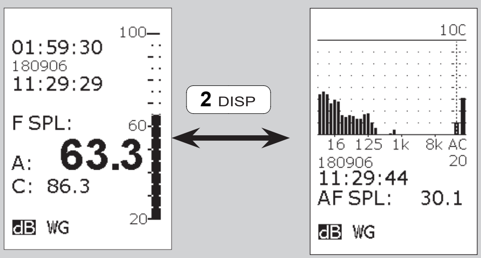
Switching between sound level meter display and spectrum display
1/1 octave display



Sound level meter display

Spectrum display

Switching between sound level meter display and spectrum display
1/3 octave display



Sound level meter display

Spectrum display

Making a frequency analysis

To start a frequency analysis measurement:

- Press the **START** key. The **R** in the display indicates that a measurement is running. The measurement is running and data acquired irrespective of whether the frequency spectrum is shown or not.

To temporarily halt an ongoing measurement:

- Press the **PAUSE/CONT** key.

To resume a paused measurement:

- Press the **PAUSE/CONT** key again. Upon resuming the instrument will go on measuring until the total measurement time elapsed equals the pre-set duration. Observe that data acquired the ten seconds immediately preceding the pause will be erased because of the back-erase function (see below).

To terminate an ongoing measurement:

- Press the **STOP** key.

To resume a terminated measurement:

- To resume a terminated measurement press the **PAUSE/CONT** key. Upon resuming the instrument will go on measuring until the total measurement time elapsed equals the pre-set duration. When a terminated measurement is resumed, the back-erase feature (see the section Measuring Sound) will not be activated.

To display other functions measured:

- Use the **FUNC** key. For the German-speaking markets these functions will include L_{eqI} and T_{Max5}

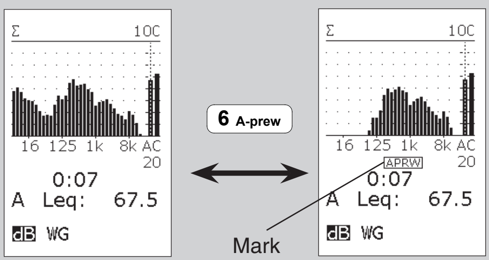
To switch between sound level meter display and spectrum display:

- Press the **DISP** key

To make the spectrum appear A-weighted

- Press the **A-PREW** key. The display will now appear A-weighted. This is purely a display function and it has no affect whatsoever, on the measured data.

A-weighted spectrum



Normal Spectrum display

A pre-weighted Spectrum display

To move the graph cursor about the frequency bands

- Use the ◀ and ▶ cursor keys. Use the ⏮ and ⏭ keys to move to the extreme left and extreme right.

To adjust the display top scale:

- If the bar graph fails to match the level measured use the **INC** and **DEC** keys to alter the display top scale setting.

Displaying the result tables

You may display all the measured data in tables. Press the **TBL** key to enter table mode. Note that the look of the tables depends on whether you start from sound level meter display mode or from frequency spectrum mode.

The tables are available during, as well as after a measurement. Remember that SPL values are shown during the measurement only.

The tables available are shown in the side bar *Displaying the result tables.*

Storing the acquired data

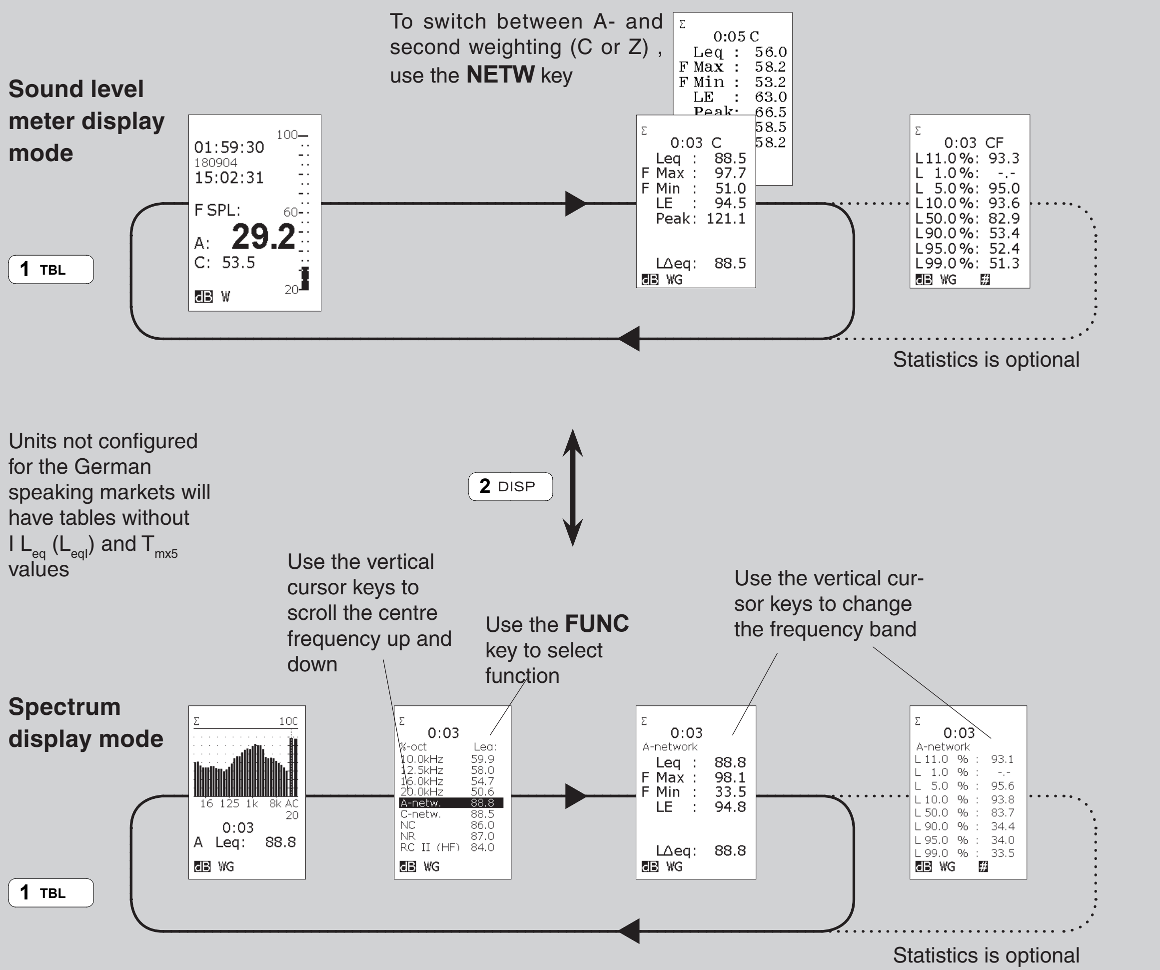
The Nor131 and Nor132 have a large, non-volatile memory to hold the result of the measurements. When the instrument is equipped with the optional Octave-band analysis, all measured functions for every octave-band are also stored together with the rest of the measurement. As for the frequency-weightings, the statistical percentiles are stored – for every octave-band.

All aspects of the memory handling are discussed in detail in *Memory handling*.

The L_{PEAK} and T_{Max5} are not measured for the octave-bands!

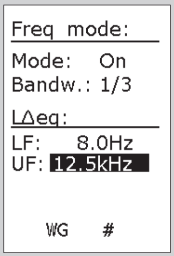
Displaying the result tables

This is valid if no selection is done for parameters in “User defined table).

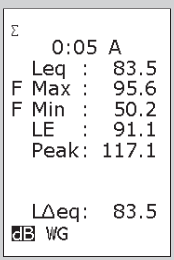


Finding the energy level over a part of the frequency range

Sometimes, the result you are looking for is the energy in a specific frequency range, but the total level is disturbed with noise coming from another source. The A, C or Z value is therefore not correct for your application. For such cases there is a special function available.



Here you can specify the upper and lower frequency of interest and the calculated Leq value for this frequency range will be displayed in the tabular result picture as LΔeq.



NC, NR and RC rating

Many applications, particularly in the heating and ventilation industry, require the use of noise rating criteria that compare the frequency spectra of the measured noise levels against the reference curves.

The NC – Noise Criteria, NR – Noise rating and RC – room criteria are supported in the Nor131/132, all returning a single dB value based on frequency rating curves. The noise rating criteria requires that the instrument is equipped with the 1/1 octave option. The rating is also calculated if the measurement is performed with 1/3 octave spectra. In this case the Nor131/132 is automatically recalculating the 1/3 octave values to 1/1 values prior to the noise rating calculation

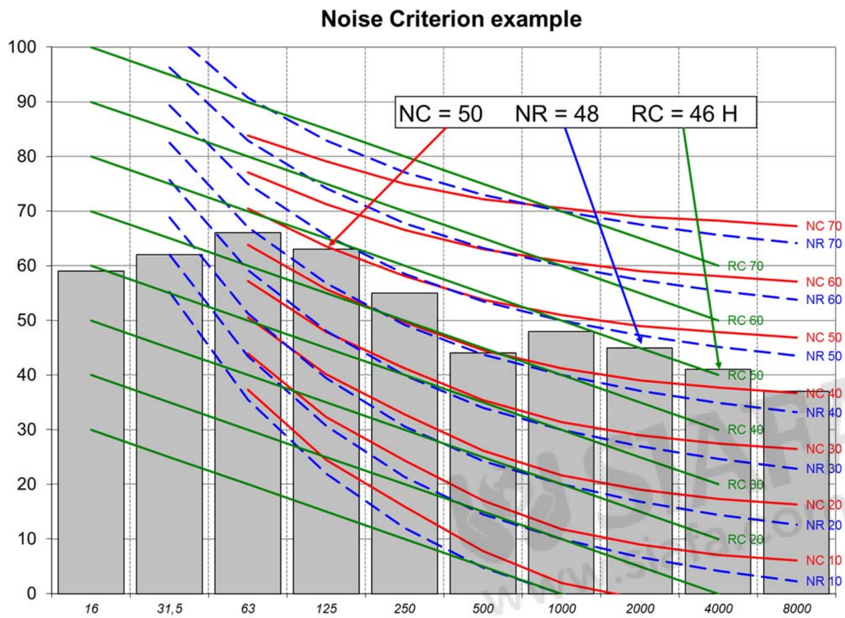
Noise Criterion - NC - were established in U.S. for rating indoor noise and noise from air-conditioning equipment etc. In Europe it is more common to use Noise Rating Curves – NR. The method consists of a set of criteria curves extending from 63 to 8000 Hz for the NC rating and from 31.5 to 8000 Hz for the NR, and a tangency rating procedure. The criteria curves define the limits of octave band spectra that must not be exceeded to meet occupant acceptance in certain spaces.

Similar the RC – room criteria value is based on a set of criteria curves in the frequency range from 16-4000 Hz. The RC value however is only based on the three octave bands 500 Hz, 1 kHz and 2 kHz. In addition is H (Hiss) and or R (Rumble) added to the value. The R is added if any 1/1 octave band between 16 Hz and 500Hz is crossing the determined RC rating curve. Similar a H is added if the 2000 or 4000 Hz 1/1 octave band is crossing the determined RC curve.

The rating value (NC, NR and RC) is obtained by plotting the octave band levels for a given noise spectrum - the rating curves. The noise spectrum is specified as having a rating same as the lowest rating curve which is not exceeded by the spectrum. The rating evaluation is performed in 1 dB steps.

The rating values are found in the bottom of the frequency spectrum table. Push the TBL button in the frequency display to get access to this table.

The noise criterion evaluation search the highest 1/1 octave band that fits below the applicable noise criterion lines. The evaluation is performed in 1 dB steps. 1/3 octave spectrums are re-calculated into 1/1 octave spectrums prior to evaluation.



Basic time profile measurements

Instruments equipped with the optional option 3, level vs. time will be able to log the time profile like the classic level recorders used to do.

The time profile is measured by dividing a total measurement into smaller periods of time, all having the same duration. Option 3 allows the period length to be from 1 second and upwards in 1 second steps. Each value is stored in the instrument memory, and at the end of the measurement the logged profile is stored together with the overall Global Measurement.

The logged profile may be transferred to a PC for further analysis. Software like Nor1026 NorReview may be used for extensive analysis of the result. A view of the profile is not available in the instrument display itself.

Global vs. profile

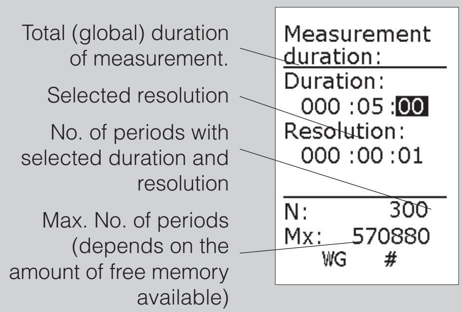
The traditional sound level measurement and the frequency analysis both consider the entire measurement as a whole without dividing it into smaller parts. One may therefore consider these two measurements as global measurements, while the level vs. time measurements represent the *profile*.

The L_{Aeq}, LAF_{max} and L_{Cpeak} are measured for every period separately and stored in a memory buffer as a profile measurement.

Profile measurements may be made in parallel with global frequency analysis and in parallel with the traditional sound level measurement.

This chapter deals with profile measurements only. For details on global frequency analysis see the chapter *Frequency analysis*.

The measurement duration setup menu



The profile resolution does not have to be selected so that the global duration becomes a multiple of the profile resolution. The last period will be truncated if the duration divided by the profile is not an integer.

Which resolution should you use?
This will always be a trade-off between the need for information and the amount of data generated. You should also take into consideration the global duration of your measurement.

For example, you may want a higher resolution in a 1 minute measurement than in a 24 hour measurement. Will you need a 1 second resolution for 24 hours? It's going to be a lot of information to go through afterwards.

No absolute rules or guidelines can be given since there are so many different applications and requirements.

Making measurements

All you need to do to set up the Nor131/132 to expand the measurements to also include the time profile is to define the time resolution.

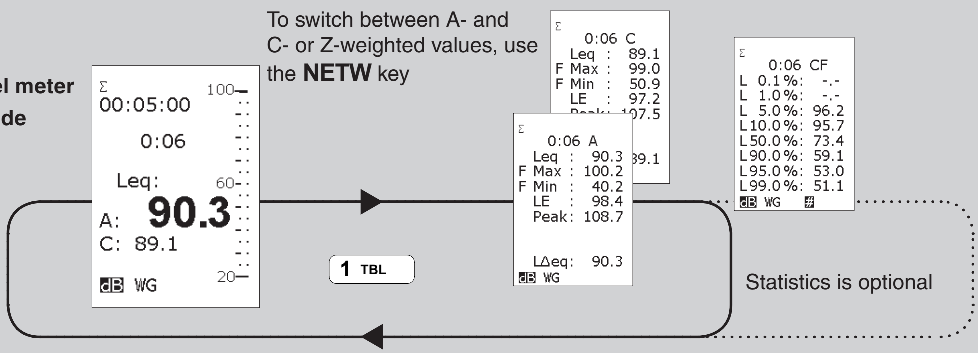
To define the duration and the resolution:

1. Press **SETUP > 2**. Units with the option 3 installed will then produce the measurement duration menu. See side bar for details.
2. Set the global duration.
3. Move down to resolution and set as required. Make sure that number of periods, N, is below the maximum value. Navigate and leave the menu as usual.

If you've set up a profile resolution different from (i.e. shorter than) the global measurement duration, the instrument will log the time profile in addition to the global measurement.

Displaying the result tables of the Global measurement

Sound level meter display mode



Units configured for the German speaking markets will have tables with I Leq (LeqI) and Tmx5 values

The effect of pressing STOP before resuming. If you terminate an ongoing measurement prematurely by pressing the **STOP** key and later resume the measurement by pressing **PAUSE/CONT** an S marker will be added to the period within which the **PAUSE/CONT** key was pressed – i.e. we mark out the first period after resumption.

Functions measured in the basic time profile mode. In the basic time profile mode, the instrument logs the A-weighted equivalent level, the A-weighted maximum sound pressure level and the Z- or C-weighted peak level.

At the same time, the global mode measures the instantaneous SPL, the maximum and the minimum SPL, the equivalent level, the sound exposure level and the maximum peak level. All the global levels are measured as A-weighted and Z- or C-weighted levels.

Reverberation time measurements

The optional extension 5 for the Nor131/Nor132 permits measurements of the reverberation time. The reverberation time is simultaneously measured in every frequency band.

Reverberation time is not something you measure directly, it is a result calculated from a measured response.

The calculation algorithms used complies with the requirements set by ISO 354: 2003 Acoustics – Measurement of sound absorption in a reverberation room and ISO 3382:1997 Acoustics – Measurement of the reverberation time of rooms with reference to other acoustical parameters. Typical applications include the measurement of reverberation time as required in the International standards for building acoustics given by the ISO 140 and ISO 16283 series.

The instrument calculates the reverberation in the range 63 Hz to 8 kHz when 1/1-octave bandwidth is selected, and in the range 50 Hz to 10 kHz for 1/3-octaves.

What is reverberation time?

Assume that you switch on a sound source in a room equipped with a microphone system. You will note that the sound level will not reach a steady level immediately. The reason why is that the sound will consist of the direct sound radiating from the source in combination with reflected sound and these reflections take time before they reach the microphone.

If you now switch off the sound source, the sound will take some time to decay. For most rooms without significant echoes, the level will decay linearly with the time until the noise floor is reached. The time it takes for the sound pressure to decay by 60 dB is called the reverberation time. However, the calculation in Nor131/Nor132 is based on the decay in the range 5 dB to 25 dB below the stationary level and extrapolated to obtain the 60 dB range value. Reverberation times measured this way are often denoted T20. In a similar way T30 is based on the decay in the range 5 dB to 35 dB below the stationary level. The instrument Nor131/Nor132 measure both T20 and T30 simultaneously based on the same excitation.

Two methods of measuring decay curves are described in the referred International Standards: The interrupted noise method and the integrated impulse response method. Nor131/Nor132 applies the integrated impulse response method.

The integrated impulse response method is often called Schroeder method after Dr. Manfred Schroeder who disclosed the theoretical relations between this and the interrupted noise method. The reverberation time is extracted from the decay by the use of a least mean square fit algorithm.

The integrated impulse response of a room is a deterministic function and not prone to statistical deviations, so no averaging is necessary. However, for many applications you still need spatial averaging, as described in the measurement standards

Calculating the reverberation time

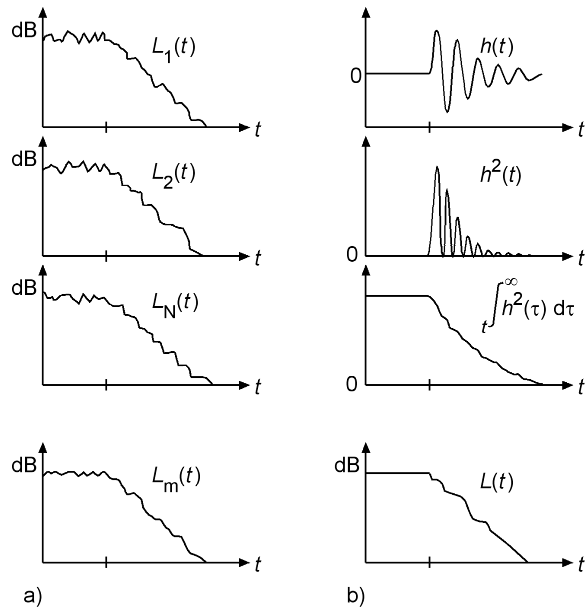
Measurement with impulse excitation

M.R. Schroeder [1] has shown that the expected decay in one particular observation point may be obtained without averaging by processing the impulse response between the excitation signal (loudspeaker) and the observation point (microphone) directly. This holds as long as the system is linear and time-invariant.

The measured response in the classical method based on noise excitation may in theory be described as a convolution between the excitation signal and the impulse response of the room. However, in the classical method with noise excitation the response is recorded directly and information about the impulse response is normally not known.

According to the Schroeder methods, the results may be obtained from processing of the impulse response itself.

Calculation of reverberation time: a) based on the “classical” interrupted noise method and b) based on impulse excitation. For the classical method the mean decay may be obtained by averaging more decays. By the use of the impulse method the expected decay is calculated from the impulse response for the room.



When a room has been excited by stationary white noise for a time sufficient to obtain stationary conditions and the noise is thereafter switched off at the time $t = 0$, the expected level at any time $t \geq 0$ will be [1]:

$$L(t) = 10 \times \lg \left[\frac{W_0}{C_{\mathrm{ref}}} \int_t^{\infty} h^2(\tau)\,\mathrm{d}\tau \right] \quad \mathrm{dB}$$

where

W_0 is a constant specifying the signal power per unit bandwidth of the excitation signal;

h(t) is the impulse response; and

C_{ref} is an arbitrary selected reference value for the level calculation.

The decay corresponds to the expected decay based on the classical method, which conventionally is approximated by a straight line.

Due to the fact that the running time, t, is the lower start point for the integration, the operation of the formula in the equation may be described as backward integration. In an alternative form of the formula, the integral starts at ∞ and runs backward to the actual time. Historically, this was achieved using analogue technology by playing a tape with the recorded response in the reversed direction.

The formula does not consider the extraneous noise normally accompanying a measurement.

When a fractional-octave-band filter is a part of the measured system, the formula will describe the expected decay according to the classical method for the applied filter band.

The difference between the classical and the Schroeder method may be illustrated by the figure on the adjacent side. For the classical method, a), the averaged decay is obtained by averaging a number of measurements. For the Schroeder method, b), the expected decay is obtained by processing of the impulse response h(t).

Implementation in Nor131/Nor132

The instrument can be set to measure reverberation times with various maximum lengths. See section below. The following description is valid for the maximum length default selection of 4 seconds.

Impulse excitation. When the instrument is set up for measurements of reverberation time based on impulse excitation, the instrument starts logging the level in each octave or 1/3-octave band with a time resolution of 5 ms. Each sample will represent the Leq-value for each 5 ms. period (200 level values per second). The sequence of these samples are stored during the measurement and processed as soon as the measurement is ended. The backward integration is started from the cross point between the decay and the background noise. The back-integrated curve can now be obtained with a time resolution of 5 ms for each filter band. Samples between 5 dB and 25 dB below the maximum are used for the calculation of T20 by application of regression. In a similar way values between 5 dB and 35 dB are used for calculating the T30 value. A least-square-fit regression method is used for fitting a linear decay curve which is used for the reverberation time calculation. The instruments automatically compare the lowest levels for the calculation with the estimated background level and gives warnings if the distance is not sufficient.

Ref.[1] Schroeder, M.R., “New Method of Measuring Reverberation Time”. J. Acoust. Soc. Am., vol. 37 (1965) pp. 409 – 412.

Excitation signals

Impulse excitation. Excitation for the integrated impulse response method may be any impulsive, broadband source with suitable low directivity. This may be a pistol shot, an exploding paper bag or an exploding balloon capable of creating enough sound energy in the frequency range you need.

An advantage of using impulse excitation is that you don't have to carry heavy loudspeaker/amplifier combinations with you. In addition you will be able to rely on battery operation alone with no need for mains voltage. A benefit of using noise excitation through a loudspeaker is that you will more easily verify proper levels and directional characteristics of the source.

Minimum reverberation time possible

The frequency analysis in the form of 1/3- and 1/1- octave filters sets a lower boundary for the reverberation times that can be measured. All frequency selective devices such as a filter will have a response shortly after removal of an input signal. This will create a virtual reverberation time and the instrument can not measure below this limit. If the measured reverberation time is below the lower limit for reliability, the value will be marked by an adjacent question mark. The table lower limits for a reliable measurement as a function of frequency and bandwidth. The values of these virtual reverberation times are increased slightly to cover for strong fluctuations in the decay curves.

Maximum reverberation times

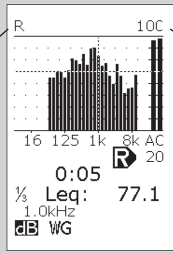
The longest reverberation time it is possible to measure is a function of the sampling speed of the reverberation curve. The default selection gives a maximum reverberation time of 4 seconds with a sampling rate of 5 ms. Alternatively you can select 8, 16 or 32 seconds. The sampling interval will be increased accordingly from 5 ms to 40 ms. Please note that this will not necessarily have a negative impact on the resolution and accuracy of the reverberation time calculation where the least square fit method is used.

Minimum reverberation times

Frequency	Lower limit 1/3-oct	Lower limit 1/1-oct
50 Hz	0,60	
63 Hz	0,48	0,24
80 Hz	0,38	
100 Hz	0,30	
125 Hz	0,24	0,12
160 Hz	0,19	
200 Hz	0,15	
250 Hz	0,12	0,06
315 Hz	0,10	
400 Hz	0,08	
500 Hz	0,06	0,03
630 Hz	0,05	
800 Hz	0,04	
1 kHz	0,03	0,02
1.25 kHz	0,02	
1.6 kHz	0,02	
2 kHz	0,02	0,01
2.5 kHz	0,01	
3.15 kHz	0,01	
4 kHz	0,01	0,01
5 kHz	0,01	
6.3 kHz	0,01	
8 kHz	0,01	0,01
10 kHz	0,01	

The level above which trigger will take place (provided that a level transition takes place) is shown as a horizontal dotted line. This line is always located 30 dB below the display top scale. To “move” the line up and down (what you do is to change the display top scale) use the **INC** and **DEC** keys. The position of the graph cursor (the frequency cursor) determines the frequency band to be used as trigger band

The instrument is in Reverberation time mode



Display top scale

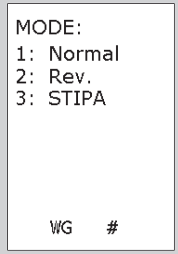
Measuring according to the integrated impulse response method

The instrument has to be set in a special mode of operation in order to measure the reverberation time. Press **SETUP > 9 > 2** for Reverberation. Select the frequency mode “On” and the appropriate filter bandwidth – 1/1- or 1/3-octave (see the chapter Frequency analysis for details). Press the **SETUP** button to change the set up for the reverberation mode.

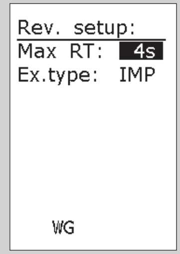
The letter **R** in the upper left corner of the display indicates that the instrument now is in reverberation time mode. By default, the cursor is located on the 1 kHz frequency band. The cursor position determines the frequency band used to trigger the measurement.

If 1 kHz is not suitable as trigger band – which may well be the case – use the cursor keys to move the cursor in the usual manner.

Reverberation Menu



Reverberation Setup



You will note a dotted horizontal line located 30 dB below the displayed top scale. Remember that the displayed top scale is purely a display feature and not related to the measurement range setting. This line represents the trigger level threshold for the impulse decay capture.

Use the **INC** or **DEC** key to change the displayed full scale and thereby the trigger level threshold in 10 dB steps, if applicable.

Press the **START** key. The instruments start logging the level and waits for the trigger condition to be fulfilled. While waiting, a **W** (waiting) is displayed. Once the instrument detects a level above the threshold in the frequency band you have specified, the capture of the impulse and its decay will begin. An **R** will appear in the display when the measurement is running, and measurement time is counted down.

You should make the impulse excitation as soon as the W mark is displayed.

Once you have pressed the **START** key the trigger setting cannot be changed, even if the measurement is not yet running. To be able to change these settings you must terminate the ongoing measurement. To terminate an ongoing measurement you have to press **STOP** or **PAUSE/CONT**. In both cases the instrument will return to ready condition. If a measurement has been made and the result table is displayed, press **EXIT** twice to enter this condition.

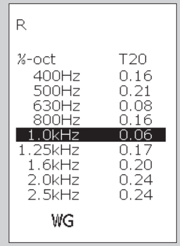
During the measurement, the instrument will show the remaining measurement time. The total measurement time is 5 seconds for 4 seconds maximum reverberation time selection, and 10, 20 and 40 seconds for 8, 16 and 32 seconds maximum times respectively. If you are measuring very short reverberation times you may terminate the measurement as soon as the background noise level has been reached by pressing **STOP**.

The calculation of the reverberation time is automatically performed as soon as the measurement is terminated and the values are shown in a table.

The table shows the calculated reverberation time for each frequency band and for the spectral weighting networks (A- and C- or Z-weighting). If the reverberation time measured is too short compared to the values in the table for minimum reverberation times, a question mark will be shown to the right of the value. If the signal-to-noise ratio is insufficient for calculating the reverberation time, the sign “-.-“ will be displayed instead of a value. An overrange value is marked by *.

Press **FUNC** for displaying T20 or T30 as appropriate.

Reverberation Table



Memory handling

The instruments Nor131 and Nor132 have a large built-in, non-volatile memory which can hold large amounts of measured data and measurement setups. These data can be transferred to a remote PC for further processing.

Memory structure

The memory structure of the Nor131/Nor132 is quite similar to that of a PC. They both have folders and files. However, as simplicity in operation is a keyword in the Nor131/Nor132 memory handling, you don't have to give the folders available for storage a name. It is automatically given the name of today's date. Neither needs you to give the files a name: They are automatically numbered consecutively in ascending order as they are stored, starting at 0001. After a storage, the picked file name (number) is displayed in the heading of the display.

Storing a measurement setup

Measurement setups can be stored for future use. This can be handy feature when the instrument is used by several people or for many different tasks.

To store a measurement setup:

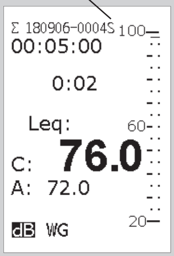
- Set up the instrument as required and press **STORE** without making a measurement. Setups are stored in a separate folder called **SETUP**.
- If a measurement has been made or a result was recalled, so a result is shown on the display, press the key **EXIT** to clear the result in order to be able to store a **SETUP.**

All settings are stored, but upon recall of a setup all settings affecting the hardware is not read back into the instrument. Hardware settings such as preamplifier gain and calibration sensitivity are examples of settings not read back.

However, all settings of functions and parameters used in the measurements are read back.

When data is stored

Folder and file number of data stored.
The S indicates Stored.



Storing a measurement

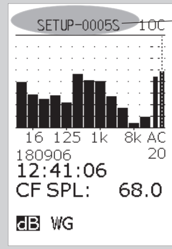
Once a measurement has been made, it can be stored in the non-volatile memory for future use.

To store the data:

- Press the **STORE** key after a measurement.

The data will now be stored in a folder with the name of today's date. If this folder doesn't exist, it will be created by the instrument. The first file gets the number 0001, the next gets the number 0002 etc.

If you choose to delete one of the files already stored you will leave a gap in the file list. This gap will not be filled with a file stored later, but be left open. Otherwise, you will easily loose track of which file contains what.



This is setup No. 5 stored in this instrument (S for stored)

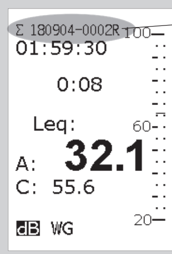
Retrieving stored setups and data

Measurements stored are easily retrieved.

To retrieve a stored setup or stored data:

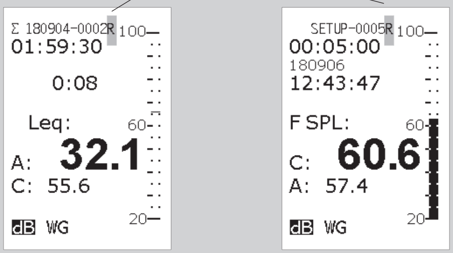
1. Press the **RECALL** key.
2. Follow the procedure explained in the side bar.

If you retrieved a stored setup or measurement this is now available for use. The fact that you have retrieved something from the memory is reflected in the text line appearing at the top of the display – see Fig.



This measurement has been stored as file No. 2 in today's folder (S for stored)

The uppermost text line in the display which file has been retrieved. The little R denotes Recalled, just like S denotes Stored



Clearing files and folders in the memory

To delete files and folders in the directory:

- Press the **DEL** key. The display will now produce the clear file menu.

In order to successfully locate the files and folders you want to delete, you must apply the procedures discussed in *Retrieving stored setups and data* (including the side bar on this page).

Clearing a single file

To clear a single file:

- Make sure that the file to be cleared is selected, i.e. highlight (shown as white text on a black background). Press the **ENTER** key. You will now be prompted to confirm your action. However, as default the cursor is positioned on the CANCEL field to avoid erasing the wrong file.
- Use the cursor keys to move the cursor to CUR.FILE and press **ENTER** again. The file is now deleted.
- Press **EXIT** if you want to leave the menu without deleting any file.

Clearing folders or the entire memory

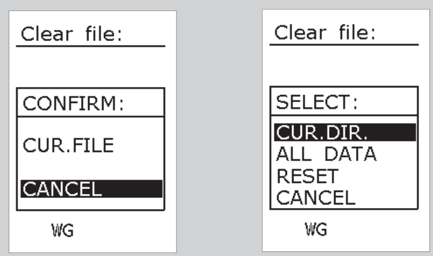
To clear a folder:

- Select the folder using the cursor keys and press enter. You will now be prompted to select between clearing the CUR.DIR. (i.e. the current folder or directory, all data in the entire memory or to reset the entire memory. The option CANCEL is also included to avoid unintended actions. If so, use exit to leave the menu.

You cannot delete the file that you are displaying. Therefore, in order to clear this file press exit before you enter the **DEL** menu.

If you select to delete all data, all measured values will be deleted, but the setup information will be retained.

The menus for clearing files, folders and the entire memory

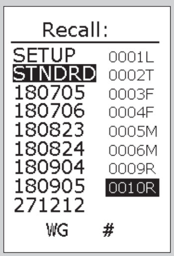


Standard set-up

Some standard set-ups are delivered with the instrument. You may use one of these set-ups as a starting point for making your own version, or simply use them as they are. The standard set-ups are placed in a folder marked “STNDRD” in the internal memory, see figure below. A description of the different set-ups is found in the table on the next page. A short description of these different set-ups is found in the table on next page.

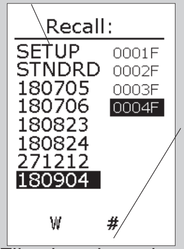
If a set-up for a particular mode is loaded, the instrument is automatically set to the corresponding mode of operation independent of the previous mode.

Standard set-up



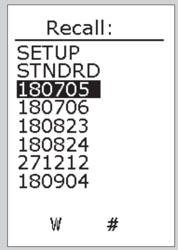
Retrieving stored setups and data

Folders

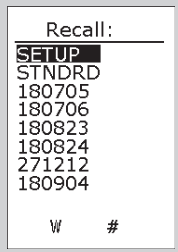


Files in selected folder

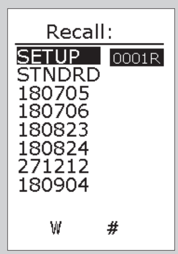
Once you’ve pressed the **RECALL** key, the display will show a list of folders and the contents of one of them (here this is the folder 150128). Use the vertical cursor keys to move up and down in the file list of this folder.



To be able to scroll in the folder list, press the ◀ key once and then use the vertical cursor keys to move to the wanted folder.



For example the folder containing all the setups.

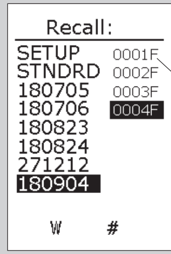


To display the files contained in the selected folder, press the ▶ key and locate the file in question by means of the vertical cursor keys.
Press **ENTER** to recall the located file/setup and **EXIT** to leave the menu without recalling any file/setup.

Standard set ups for the different sound level meter modes	
Name:	**Description / Use:**
0001L	**Normal sound level meter mode measurements**
0002T	**Normal level versus time mode measurements**
0003F	**Simple frequency analysis with 1/1 octave resolution**
0004F	**Simple frequency analysis with 1/3 octave resolution**
0009R	**Reverberation time measurements in 1/1 octaves with impulse excitation**
0010R	**Reverberation time measurements in 1/3 octaves with impulse excitation**

NOTE! The reverberation time setups 9R and 10R specify impulse excitation from an external source, such as an impulse from a pistol shot or bursting paper bag.

Keeping track of the measurement mode the file was stored in. The different modes are indicated in the file list as follows:



All file names (i.e. file numbers) have a letter as suffix. This letter indicates the measurement mode:

- **F** means frequency analysis, but no profile.
- **L** means that the file contains a simple global measurement, but no frequency analysis and no profile.
- **P** means a measurement made in sound power mode.
- **R** means a reverberation time measurement.
- **T** means a profile measurement with or without a frequency analysis.
- **$** means a STIPA-measurement.

Noise monitoring

Due to its large memory and the high dynamic range, the Nor131 and Nor132 are well-suited for unattended noise monitoring applications. Some installations, semi-permanent or permanent, are based on tight computer control, while others leave more of the job to the measuring instrument itself. The instruments can be used with success in both types of systems.

For outdoor monitoring, the Nor131 is recommended since its detachable microphone may be separated from the instrument and placed in a microphone protection system like Nor1218. See also “Powering the microphone preamplifier” on page 12 .

The Norsonic environmental solutions contain a complete range of equipment and accessories for environmental noise measurements and monitoring, all the way from outdoor microphone units, via enclosures and transmission cables to controlling and post-processing software. A detailed presentation is available on www.norsonic.com.

Automated storage of measured data

The Nor131 and Nor132 can be set up to measure for a predefined period in time and then store the measured data and start over again automatically.

The snag, however, is that a little time will always be spent on storing the acquired data. This means that if you, for example, set up the instrument to measure in periods of one hour and start the measurement exactly on the hour, the measurement period start time will exhibit a lag after some hours of measuring – typically 3–4 seconds per individual measurement.

If this lag is unacceptable to you, we recommend that you use the ***synchro*** feature. When activated, the synchro will stop the measurement a few seconds earlier to give room for data storage and housekeeping so that the next measurement will start exactly on the hour.

Available storage modes

The Nor131 and Nor132 will always operate in one of the four available storage modes. These are:

- **Manual,** which requires that acquired data are stored manually by the operator before the next measurement is made.
- **Automatic,** which causes the acquired data to be stored automatically upon measurement termination, regardless of the reason for termination – irrespective of whether termination took place because the duration expired or because you pressed **STOP**.

What can be done to the measured data?
Data acquired are available for inspection, during or after a measurement.

You may:

- Switch between sound level meter display and frequency spectrum display
- Display the functions measured
- Display the result tables
- Change the spectral weighting function Between A- and C- or Z-weighting, this depends on which one you measured
- Display the eight percentiles (requires the presence of the optional extension 4) and set one of them as you like
- Store them for future use

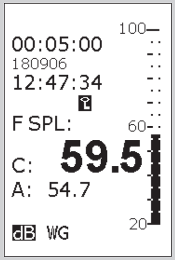
- **Repeat,** which causes the instrument to store the acquired data and then restart immediately and make another measurement using the same measurement setup and duration. Repeat applies to measurements terminated by themselves only. If you terminate a measurement by pressing **STOP**, the instrument will not restart. Note that some time will be spent on storing the acquired data, There-fore a slight delay in the restart moment will be observed. *Use at least 2 seconds measurement time if you use* REPEAT!
- **Synchro,** which compensates for the time spent on housekeeping (i.e. storage of data etc.) to maintain synchronisation with the time of day. This works in the way that the instrument synchronises itself with the full hour of the time of day. To be active, synchro requires a minimum measurement time (duration) of 30 seconds per individual measurement.

Synchro – an example

Assume that you set up the instrument to measure in periods of one hour and that you start the measure-ment at 08:52:40. The first period will be truncated and last a little less than 7 minutes and 20 seconds to give room for storage before 09:00:00. The succeed-ing period will then each be very close to an hour long to enable restart again at 10:00:00, 11:00:00 etc.

The principle has been designed with period lengths of typically one hour, half an hour, 15 minutes etc. in mind.

Keyboard lockout – locking the keyboard to prevent unauthorized operation. You may lock the keyboard to prevent the instrument from being tampered with while it is left on its own.



To lock the keyboard:

Press |◀, ▶, ▶|, ◀ to lock the keyboard

To unlock a locked keyboard:

Press ◀, ▶|, ▶, |◀ to unlock the keyboard

Note that the instrument must show the sound level meter display for this to work (in this display the cursor keys are not used).

Setting the storage mode

To set the storage mode:

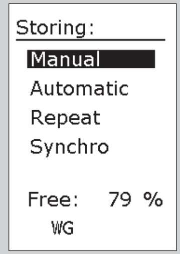
- Press **SETUP** > **1** (Instr.) > **1** (Storing). Use the cursor keys (located below the display) to navigate in the menu as usual and set the storage mode as required.

Other setup aspects

The setup for a monitoring job will depend on the task, so no absolutes can be given here. However, you should consider such things as

- What information will you need, and how detailed should it be?
- Measurement period length

Menu for storing the measurement results



- The type of outdoor microphone unit (for semi-permanent or permanent installations)
- Adaptors and cables needed (if applicable)
- Cabinet or casing required for the sound level meter
- External power to the instrument (batteries or mains connection)

The setup of Nor131/Nor132 will be found in this manual, while all the accessories can be found in separate leaflets or on www.norsonic.com.

Using windscreen? Using a windscreen reduces the unwanted noise generated by the wind, but will slightly alter the frequency response. To retain the high measurement accuracy you may switch on the windscreen correction. (**SETUP > 1 > 4 > 2**). The effect of using a windshield is discussed in Windscreen in the chapter Technical specifications.

Speech Intelligibility by STIPA

Introduction – What is STIPA

Speech transmission index – STI

The speech transmission index, STI, has shown to be a valuable tool for objective rating the speech intelligibility. From its first presentation in Acustica in 1971 the method has been refined and developed for various applications. The International Electrotechnical Commission, IEC, has launched the fourth revision of the International Standard specifying the method for calculating the index as IEC 60268-16:2011. Essential for this development was the work carried out at TNO-Human Factors in the Netherlands, and in particular the pioneers Tammo Houtgast and Herman Steeneken.

The STI-methods can be used to compare speech transmission quality at various positions and under various conditions within the same listening space; in particular it is useful for assessing the effect of changes in acoustic properties. This includes effects from the presence of an audience or of changes in a sound system. The methods are also able to predict the absolute rating of the speech transmission quality with respect to intelligibility when comparing different listening spaces under similar conditions or assessing a speech communication channel.

Development of STI

The basis for the STI-index is that the intelligibility of speech is largely based on the slow modulation of the strength of the sound pressure signal that acts as a carrier. For the full STI-method, the carrier is a stationary gaussian noise signal divided in seven bands in octave steps ranging from 125 Hz to 8 kHz. The bandwidth of each band is one-half octave. Each of the bands is modulated with 14 modulation frequencies - one by one. The modulation frequencies are selected in one-third octave steps from 0,63 Hz to 12,5 Hz. This gives in total 98 combinations.

In the STI-context, the square of the sound pressure is called intensity. The intensity is the quantity being modulated. A small loudspeaker playing the modulated excitation signal, acts as a talker.

The sound in the listener position is received by a microphone. The level and the degree of modulation in each octave band is used to determine the speech transmission index. Noise and reverberation in the room will reduce the observed degree of modulation. The method also considers the effect of the most common types of distortions such as harmonic distortion and intermodulation. However, other forms of non-linearity, like frequency shifts and frequency multiplications, are not treated effectively.

In order to fully take care of the effects of non-linearity, it is important that the basic signal being modulated is a noise signal with a high crest-factor, a spectral distribution similar to the long-term speech spectrum, and that the main modulation frequency is selected one by one. The measurement of the full STI therefore has to be performed as a sequence of measurements. If each of the 98 combinations is measured for 10 seconds, the total measurement time will be about a quarter of an hour. The need for such a long measurement time in order to obtain the STI-value in one position of a room limits the applicability of the full STI-method.

The STI-method may be modified in different ways to reduce the time needed for the measurement. If the system to be measured is regarded as linear, then a number of solutions exist. The excitation signal may be modulated with all modulation frequencies simultaneously and the components may be separated after reception by the use of filters or Fourier analysis. A more common method is to calculate the complex modulation transfer function from the impulse response of the room.

If the impulse response can be regarded as a well-behaved room response with an exponential decaying envelope characterised by the reverberation time, the modulation transfer function at frequency, F, may be calculated directly from the value of the reverberation time, T, and the effective signal-to-noise ratio S/N in dB. A simplified formula, not taking the effects of masking and the threshold of hearing into consideration, indicated the following relationship:

$$m(F) = \frac{1}{\sqrt{1+(2\pi F \frac{T}{13,8})^2}} \cdot \frac{1}{1+10^{(-S/N)/10}}$$

As seen from this formula, a limited signal-to-noise ratio reduces the modulation transfer function for all frequencies. A long reverberation time reduces the modulation most for the highest modulation frequencies.

The STI-value is a weighted average of the different modulation indexes. The last revision of the method (IEC-standard) also considers masking effects and the absolute threshold of hearing.

RASTI and STIPA

In order to simplify the direct measurement, the RASTI-method (Room Acoustic Speech Transmission Index) was developed at TNO in 1979. Different instruments were developed for the measurement according to this standard. A typical measurement time was 10 to 15 seconds. The RASTI method only considers two octave bands 500 Hz and 2 kHz.

Due to the simplicity in use, the RASTI-instruments were used also for applications beyond the main design goal – room acoustics. The RASTI-value is often used for assessing the quality of public address systems, but comparisons with subjective measurements have shown that the deterioration of speech intelligibility is not handled correctly if the PA-system is strongly non-linear or suffers from limited bandwidth.

In order to improve the accuracy in the intelligibility assessment, the STIPA-method was developed. It handles effects due to reverberation in the room and distortions commonly found in public address systems. It also performs well for room acoustics and can therefore in nearly all cases replace the RASTI-method and deliver results more closely to the values obtained by the full STI-method. The measurement time for a STIPA-measurement is similar to the RASTI-method: 10 – 15 sec.

The RASTI-method is in the current version of IEC 60268-16 obsolete.

The STIPA method uses a higher degree of modulation for each test frequency. It will thus be more robust for interference from non-stationary background noise. See figure below for a comparison between the methods.

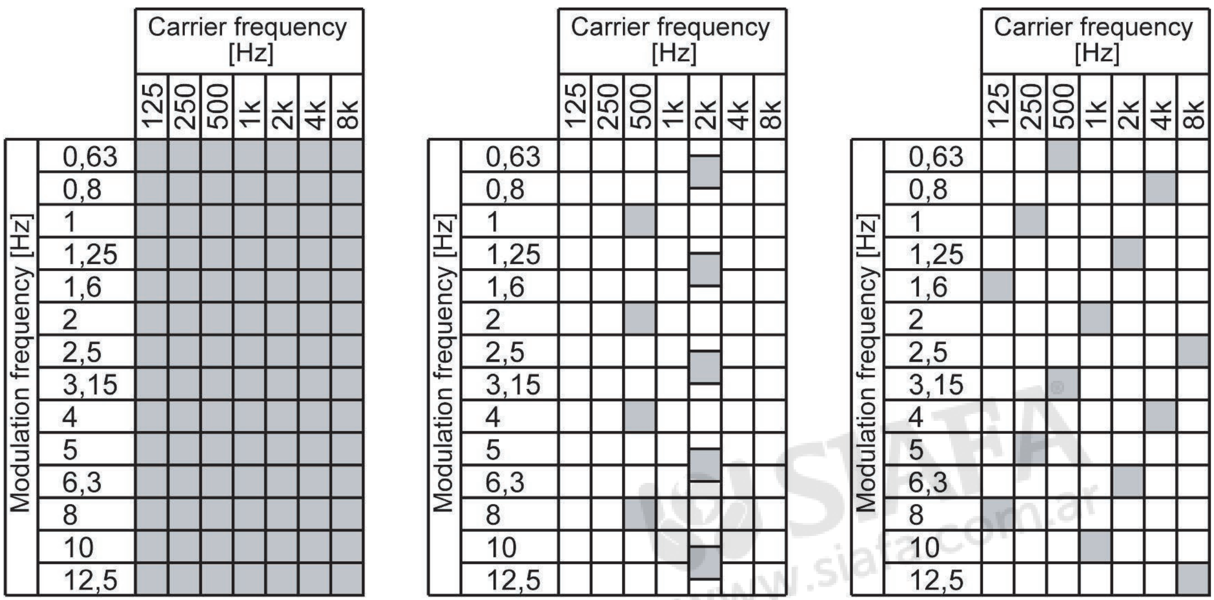
Note that the STIPA-method consider the aboslute level of the sound and mimic effects from the treshold of hearing as well as level dependent masking. Calibration of the sound level meter is therefore important.

The STI value will be lower for speech levels below 50 dB and above 80 dB.

How is STIPA measured

The instrument option comes complete with an excitation signal matched to the program option. The excitation is played continuously through a loudspeaker acting as a talker.

For a STI measurement, all carrier frequency bands are modulated with all modulation frequencies – in total 98 modulation indices. For RASTI, two carrier frequencies are used giving 9 modulation indices. For STIPA, 14 modulation indices are measured from the complete range of carrier frequencies



STI RASTI STIPA

The battery operated sound level meter is placed in the position in the room where you want the speech intelligibility to be judged. About 15 seconds after the start of measurement, the STI-value will be indicated on the screen of the instrument. No cable is needed between the excitation and the instrument.

CIS

The instrument will use the measured STI value and also present the speech intelligibility in an alternative scale called " Common Intelligibility Scale" abbreviated CIS. This scale is according to the definition in IEC 60849 Sound systems for emergency purposes. It is a non-linear relation between the STI and CIS value although both use 1 to indicate the best intelligibility and 0 for the poorest.

Units included in delivery

Software option for Nor131

The STIPA calculation program is designated as an option in the sound level meter. The option may be activated when the instrument is first delivered or installed on the sound level meter at a latter date. Please contact your local Norsonic dealer or the factory if you want your instrument upgraded with this option.

The STIPA-option requires the following additional option to be installed in the sound level meter:

- Option 1: 1/1-octave real-time filters in the frequency range 8-16.000Hz

Excitation file - a zipped file

Please download the audio file from
www.norsonic.com/downloads. Here you can download the STIPA - rev. 4 - 2011 file. Included with the STIPA-option, contains two tracks: Track 1 and 2.

- Track 1: 1 kHz sinusoidal signal.
  Duration: 2 minutes

- Track 2: STIPA excitation signal, male speech.
  Duration: 70 minutes

Both signals have the same A-weighted level. Track 1 may therefore be used for test or calibration purposes – although the noise signal is recommended.

The STIPA excitation signal consists of bands of random noise each modulated with two frequencies. See IEC 60268-16:2011 for further details. Note that if a loudspeaker is used for excitation, the loudspeaker's frequency response may change the spectrum of the excitation. The sound level meter may be used to verify or adjust the spectrum. For an excitation signal with A-weighted level Lx, the various octave bands should have the levels shown in the table on the right:

NOTE that the excitation file is matched to the version of the STIPA-option.

Excel sheet

The zipped file contains two Excel work-books to be used with a PC:

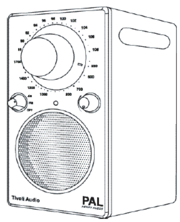
- STIPA-Calc - Allows correction of the STIPA result for a virtual background noise.

- STI-Calc - A sheet for estimating the STI-value based on the speech level, background-noise level and the reverberation time

Octave band levels for male speech signal of A-weighted level Lx						
125 Hz	*250 Hz*	*500 Hz*	*1 kHz*	*2 kHz*	*4 kHz*	*8 kHz*
Lx +2,9dB	*Lx +2,9dB*	*Lx - 0,8dB*	*Lx - 6,8dB*	*Lx - 12,8dB*	*Lx - 18,8dB*	*Lx - 24,8dB*

Octave band levels for male speech signal of A-weighted level 66 dB						
125 Hz	250 Hz	500 Hz	1 kHz	2 kHz	4 kHz	8 kHz
68,9 dB	68,9 dB	65,2 dB	59,2 dB	53,2 dB	47,2 dB	41,2 dB

Loudspeaker - not included!



A loudspeaker for the excitation is *not included* in option 13. If you need advice for selecting suitable devices, please contact your local Norsonic representative.

Norsonic can deliver a commercially available small radio with rechargeable battery: Tivoli Audio PAL. It comes with a jack-socket for the excitation signal from the CD-player. When ordered through Norsonic, you will also receive a mounting bracket for mounting the radio on a tripod. The diameter of the loudspeaker is about 6 cm and the frequency response is fairly flat from 100 Hz to above 10 kHz.

Playing the excitation file

It has been reported that the selection of the player for the excitation file may influence the measurement. We therefore recommend testing new equipment before you start the normal measurement tasks. If you supply the excitation signal electrically to the sound level meter (BNC- Lemo adaptor or cable available), or measure close to the loudspeaker, you should obtain excellent speech intelligibility ratings with a STI-value close to 1.

If you play the file from a PC, ensure that signal processing features like volume leveller and surround virtualizer are turned off. Such processing may limit the maximum STI-value and lead to unaccurate measurements.

Measurement overview

General instrument description

The figure shows a typical set-up for a STIPA measurement. The sound source, normally a small loudspeaker is acting as a talker. The loudspeaker should be placed in the normal position for a talker, either using or not using a public address system as required. The level should correspond to the normal level for speech.

The signal to the loudspeaker is obtained by playing the supplied audiofile with the recorded excitation signal.

Place the sound level meter, with the STIPA option installed, in the place where you want the speech intelligibility to be measured. Press **MODE** and select the STIPA-mode. Press the **START** button. After about 15 seconds the estimated STI-value and the corresponding CIS value is indicated on the screen.

Selecting a loudspeaker for the excitation

Most applications of the STIPA-method require a loudspeaker to act as a talker.

In this case, the directivity of the loudspeaker should be close to the directivity for a real human speaker as speech intelligibility depends upon the directivity of the source; therefore, a mouth simulator having similar directivity characteristics to those of the human head/mouth should be used for the highest accuracy when assessing the intelligibility of non-amplified talkers. Further, the frequency response between 80 Hz and 12 kHz should be flat. The loudspeaker should be a single element design or using coaxial element so the acoustical centre is well defined.

IEC 60268-16 recommend using a loudspeaker with a cone diameter of maximum 100 mm and refers to ITU-T Recommendation P.51 describing an artificial mouth. A sound source according to this specification is available from different manufacturers, with a typical example being the GRAS-44AA.

When speech is relayed through a sound system, a simulator is not normally required unless a close talking or noise-cancelling microphone is involved.

Loudspeaker or electrical excitation

Most applications require that the excitation is performed using a small loudspeaker acting as a human talker. However, same applications, such as testing public address or voice-alarm systems, are more conveniently tested with the electrical signal from the excitation file fed directly into the system to be tested.

In a similar way, some applications related to test of transmission channels require an electrical signal as the input to the sound level meter for analysis. For this application, the microphone preamplifier may be substituted with a cable to the input socket of the sound level meter.

NOTE! For this description, a measurement set-up with acoustic excitation and detection using the normal measurement microphone is assumed!

Performing a measurement

Set-up

Set the sound source (artificial mouth or suitable test loudspeaker) in the normal position for the speaker. If a sound system is being used, place the sound source on the axis of the appropriate microphone at the normal speaking distance (measured from the lip-circle for the artificial mouth or acoustic centre of the loudspeaker) and direct it in the normal speaking direction. Connect an appropriate audio-player to the sound source.

Set the speech level

Switch on the sound level meter and ensure that it is properly calibrated. Play track 2 with the STIPA excitation signal and adjust the level to the required speech level, normally 60 dB at 1 m from the sound source. (66 dB at 0,5 m) Alternatively, track 1 with the sine excitation may be used – both should produce about the same A-weighted level.

After setting the speech level, play the STIPA excitation signal on Track 2 during the measurement. (70 minutes).

The standard IEC 60268-16 recommends that the excitation spectrum is correct within ± 1 dB for the applicable frequency range.

Select the STI-mode

The STI mode is one of the modes of operation for the sound level meter. By selecting this mode, the instrument will automatically be configured for STI-measurements. For selection, press **MODE** after switching the instrument on, and select STI by pressing the numeric button **5**. After the selection, the instrument is ready for a STI-measurement. A “$”-sign in the upper left corner of the display indicates the STI-mode of operation.

The instrument screen will show the level versus frequency display. Time constant will be F. The cursor will be positioned for indication of the A-weighted level.

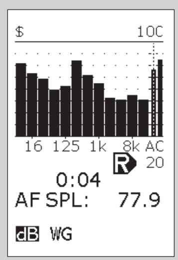
Making a measurement

Start a measurement by pressing the **START** button. The measurement lasts for 13 seconds. During the measurement the level versus frequency display will be shown together with an indicator of an ongoing (running) measurement. After the measurement is ended, the main STI display will be shown after a short calculation period.

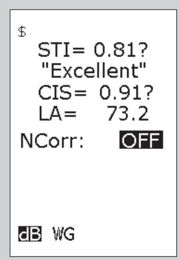
For the first measurement after the STI-mode is selected, the Noise correction will be selected off and the cursor will be placed above the “Off” field.

The measured result will be indicated as a STI-value together with an assessment of the intelligibility. The STI-value and the assessment are according to the STIPA-procedure in IEC 60268-16 for male voice. The corresponding CIS-value (Common Intelligibility

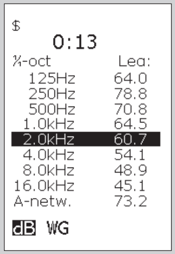
Display during measurement of STIPA



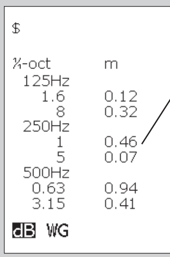
Main result display



Display of speech level octave-band spectrum



Two modulation indices are measured for each octave-band



Modulation index for 500 Hz octave-band, modulated with 3,15 Hz.

Scale) according to IEC 60849 is also indicated. The mean A-weighted speech level is indicated in dB relative to 20µPa.

If the measured value is detected as unreliable, a question mark is placed behind the indicated value for STI and CIS.

The speech level for each octave may be displayed by pressing **TBL**.

A second press on the **TBL** button displays the modulation indexes for each of the STIPA-octaves.

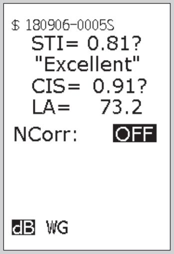
The number below each octave frequency is the modulation frequency in Hz. The corresponding un-corrected modulation index is displayed to the right. Press the key **TBL** again to display the highest frequencies. Press the **TBL** button once more for return to the main display.

NOTE! We recommend that from time to time you test your audio-player, loudspeaker and sound level meter under close to ideal conditions. Such a situation exists just in front of the loudspeaker placed in a large room when the background noise is considerably lower than the excitation signal. Adjust for a level between 60 and 70 dB. In this case, you should obtain a STI-value close to one.

Store and recall the result

After the measurement is performed, the results may be stored in the non-volatile memory of the instrument or on the SD-card; this is done by pressing the button **STORE**. Alternatively, the instrument may be set up to store automatically after each measurement, see the section “Memory handling”. A stored result may later be recalled.

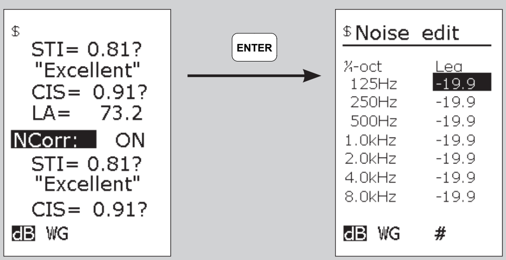
When stored, a file number is shown in the upper line, is automatically assigned to the measurement



Correcting for background and occupancy noise

The STIPA method considers the effect of the actual background noise when the intelligibility is assessed. However, in some cases we want to find out what the intelligibility would have been if we had a certain background noise. A typical application will be to assess the intelligibility in an auditorium with audience when the auditorium was measured without. For such applications, the optional noise correction may be used.

Edition of the background noise level



Measure the STI-value as described above. Place the cursor above the field “Off” in “Noise correction: Off”, and press **INC** or **DEC** button to toggle the value to “On”. The results will be recalculated to account for the specified background noise.

The result is corrected for a stored background noise spectrum. If you want to edit the values, move the field cursor to the field “NCorr:” and press **ENTER**. The field cursor is automatically placed at the lowest octave-band 125 Hz. Write in the required level for each octave by using the numeric keyboard. An “#E” at the bottom of the display indicates numeric keyboard (second function) and that each entry has to terminated by pressing **ENTER**. Move the cursor field to the next frequency by using the cursor buttons below the display and enter the corresponding level. For terminating the edition and returning to the main menu, press an additional **ENTER**. The noise corrected values are automatically recalculated.

An alternative to keying in the levels for the background noise is to use values from a previously stored measurement. The measured levels must have been made in 1/1- or 1/3-octave bands. Press **RECALL** while the table for the background noise is displayed. Select the requested file from stored results and press **ENTER** to read in the data. Values measured in 1/3-octave bands are recalculated to 1/1-octave values.

NOTE! You may obtain the averaged noise level from different measurement positions by making a normal measurement in the first position, then move to the next and press **CONT** for prolonging the measurement time!

Real background noise

If a real background noise is present during the measurement, the effect of the noise will in most cases be treated as noise according to the STIPA-method and lower the STI-value. However, some noise signals may be interpreted as a STIPA excitation signal wrongly giving a modest STI value. The instrument is programmed to detect such situations and place a question mark adjacent to the indicated value. However, not all cases can be detected properly. To investigate such potential problems, we recommend making a measurement without an excitation signal. If this STI-value is low or considerably lower than the value measured with the excitation signal (preferably STI ≤ 0,2), the reading will have a high degree of reliability.

If the background noise can be switched off, it is possible to measure the response to the STIPA-excitation and noise level separately. The STI-value may be corrected for the background noise later – in the instrument or by applying the Excel-sheet “STIPA-calc” delivered with the instrument option. This will eliminate the interference between a spiky-noise signal and the STIPA-method.

Precision of the STIPA method

Because the test signal is band-limited random or pseudo-random noise, repetition of measurement does not normally produce identical results, even under conditions of steady interference. The results centre on a mean with a certain standard deviation. Typically, the value of the standard deviation is about 0,03 with stationary noise interference. With fluctuating noise (for example, a babble of voices), higher standard deviations may be found, possibly with a systematic error. This can be checked by carrying out a measurement in the absence of the excitation signal. This should result in a residual STI value less than 0,20. An estimate of the standard deviation should be made by repeating measurements for at least a restricted set of conditions/positions.

Analysis and interpretation of the results

It is important to examine the modulation matrix to determine the reliability of the results.

As a rule, the modulation index in each octave-band should decrease with increasing modulation frequency. Constant or slightly reducing values in a column indicate the presence of noise. Large reductions indicate that reverberation is the main effect. Values that first reduce and then increase with increasing modulation frequency indicate the presence of periodic or strong reflections, which may produce an over-optimistic conclusion. It is recommended that if this effect is detected, it should be reported with the results and an estimated correction applied.

Limitations of the STIPA method

The STIPA method should not be used for those public address systems that

a) introduce frequency shifts or frequency multiplication;

b) include vocoders (i.e. LPC, CELP, RELP, etc.);

c) have a background noise that is impulsive;

d) introduce strong non-linear distortion components.

If d) applies, or possibly applies, the full STI method should be used instead or used to verify the results obtained by the STIPA method.

Using Excel for further calculations

An Excel workbook “STIPA-Calc” is delivered with the STIPA-option. This is part of the file available for download from www.norsonic.com/release. The workbook allows you to correct your readings for different speech levels and levels of background noise similar to the corrections available in the instrument.

Recall the stored results from an earlier measurement. Press the button **TBL** to display the speech level and enter the levels into the cells for the speech level. Press **TBL** once more to obtain the modulation indices and bring the values into the appropriate cells. You may then enter values for the background noise and see how this will affect your reading.

The values from the STIPA-measurement may be transferred to the PC by using the USB or serial interface on the instrument. and the file transfer program NorXfer.

STI-Calc

In the zipped file with STIPA-calc, you will also find a program called STI calc. This program may be used to estimate the STI-value based on information on the octave-band levels for speech and background noise and the reverberation of the room – also in octave bands. The applied formulas assume linearity and that the impulse response for the room is well behaved without significant echoes. The calculation should not be used as a replacement for a real measurement of the STI-value.

Specifications

The STI-value is measured according to the requirements for STIPA method defined in:

IEC 60268-16 (Ed.4.0 2011-06): Sound system equipment – Part 16: Objective rating of speech intelligibility by speech transmission index.

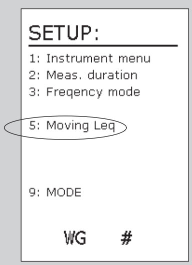
The CIS-value is calculated from the STI-value as specified in:

IEC 60849 (Ed. 2.0 1998-02) Sound systems for emergency purposes.

Moving Leq

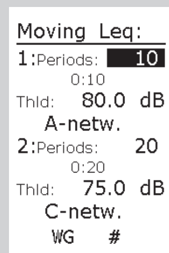
A useful addition to the measurement capacity of the Nor131/132 is to be able to follow the noise level over a certain window in time after a certain threshold level has been reached. This may eg. be an event that where there are restrictions to the noise level for a particular activity, for example a rock concert for which the Leq and Max levels shall be monitored and reported. For this purpose we have introduced a term we call "Moving Leq". In the Nor131/132 we have added this to the other measurement parameters, so it is done in parallel. We also extended this to include 2 separate user defined periods with each having their separate threshold levels, which means you can follow 2 separate events simultaneously.

To set up the parameters, go to the setup and select the function:



PS! The Moving Leq function is available for instruments equipped with Option 3 - Level vs. time.

You are then prompted to key in the duration and threshold level for Period 1 and if needed also for Period 2. Either select to follow the A-weighted levels or alternatively the Z (or C)-weighted levels

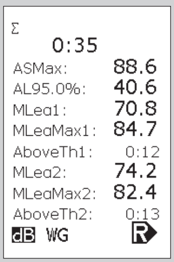


PS! If you only need to follow one event, ie. Period 1, just fill in 0 (zero) for in the field for duration for Period 2.

PS! In case you want to disable the Moving Leq function altogether, set Period1 to 0 (zero)

PS! Please note that the minimum resolution is 1 second.

During the measurement you can follow the build-up of the values by choosing the tabular display



Please note that the parameters normally found when pressing **TBL** will be replaced with the moving Leq parameters.

NB! You need have the SLM display first then press **TBL**

The displayed values MLeq1 and MLeq2 give you the Leq over the “moving windows” with the set duration while the measurement is running, in the above example 10 seconds and 20 seconds respectively. Please note that these values are not stored as they only represent the momentary values. Instead it is the MLeqMax1 and MLeqMax2 that gives you the maximum value, ie. the period with the highest Leq, In addition you also have total duration the noise level was above the set threshold values.

A marker will be inserted for each Period from when the condition is met, ie. the noise is above the threshold level. This marker can later be found in the NorReview level vs. time display and further analysed using the Event Analysis module.

Using NorXfer, the Moving Leq values are reported as follows in the Global sheet:

A	B	C	D	E
Period length	(0:0:1.0)	H:M:S.mS		
Total number of periods	61			
Number of periods before trigger	0			
Number of periods after trigger	61			
Trig time	(2014/11/5 13:58:59.0)	Y-Mo-D H:M:S.mS		
Measurement effective duration	(0:1:0.0)	H:M:S.mS		
Moving window length 1	(00,00,00,10,000)			
Network 1	A-network			
MaxMovingLeq1	115,2	dB		
PeriodsMaxMovingLeq1	38		(2014,11,05,13,59,37,000)	
PeriodsAboveThreshold1	28		(00,00,00,28,000)	
MLeqThold1	70	dB		
Moving window length 2	(00,00,00,05,000)			
Network 2	A-network			
MaxMovingLeq2	118,2	dB		
PeriodsMaxMovingLeq2	43		(2014,11,05,13,59,42,000)	
PeriodsAboveThreshold2	6		(00,00,00,06,000)	
MLeqThold2	90	dB		

LAeq / LAImin / LASmin / LAFmin / LAImax / LASmax / LAFmax / Moving Leq / Summary / Setup

Signal out

The sound level meters Nor131 and Nor132 are equipped with a signal out terminal. The signal is a replica of the microphone- or input signal. You may use the terminal for listening to the measured signal, or you may use it for recording purposes.

The gain is set via the selection of displayed full scale. Full scale on the display corresponds to 100 mV on the signal out terminal. Although the selection of full scale will not affect the measurement, it will determine the sensitivity for the signal out. Use the **INC** and **DEC** keys to adjust the gain. The gain may be varied over a range of 60 dB in 10 dB steps.

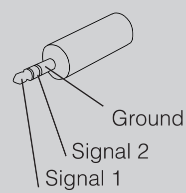
The signal out terminal can drive loads with an impedance down to less than 16 ohm, but we normally recommend a headset with 32 ohm impedance. Even a short-circuit will not affect the measurements, but should be avoided due to an excess power consumption.

The signal output terminal is a 3,5 mm stereo-jack. Both channels have the same signal, but are driven from separate amplifiers and should therefore not be connected together.

Use a stereo plug! Never use a mono plug for the signal output jack as this will short-circuit one of the outputs.

Listen to the vibrations! You may connect an IEPE-type of accelerometer to the instrument instead of the microphone. With the signal out feature, you may listen to the vibration signal. Be careful with your ears. Adjust the gain to an appropriate level.

Jack-plug for signal out



Transfer of data to a PC

To transfer measured data from the Nor131 and Nor132 to a PC you will need a USB cable (available separately, contact your local representative or the factory).

The recommended way to transfer data to a PC is by means of the software program NorXfer, (can be downloaded from www.norsonic.com/release) which includes the necessary USB driver.

The instrument can also be remote controlled. For a complete list of remote control commands contact your local representative or the factory.

Installing NorXfer

Download NorXfer from our website

www.norsonic.com/downloads.

Before reaching the site where the programs resides, you will be prompted to enter name and email, so that we will be able to contact you if this should be necessary.

Click on NorXfer and download program and follow the online instructions to install NorXfer:

NOTE! If this installation is an update of an existing version of NorXfer, the old version must be removed before you proceed with the installation itself. Observe that files stored in My Measurements are left unaffected by this.

NorXfer normally doesnot need any codes to run, unless opt.1 or 2 are installed.

Installation of USB drivers for Nor13x

The Nor13x device driver is normally distributed and installed with other Norsonic software. It may also be distributed and installed as a standalone package.

Please download drivers from:

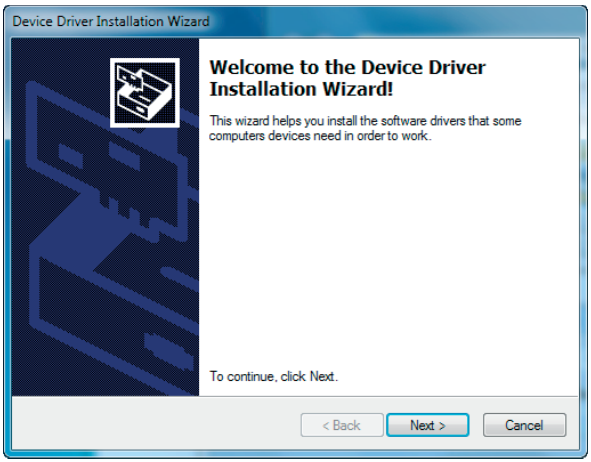
www.norsonic.com/downloads.

In case of standalone distribution, please double-click the exe-file to start the installation.

The driver is Vista, win7 and Win 8.x compatible

When the driver-installation starts, the following dialog-window will appear.

Click “Next”.

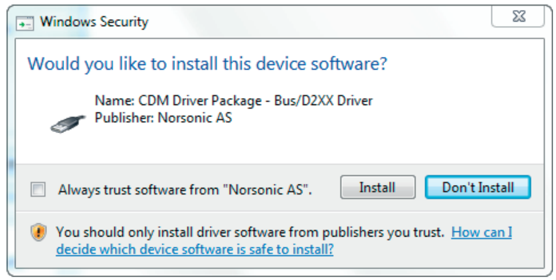


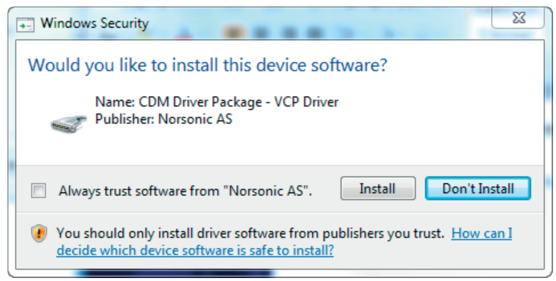
The next two dialog-windows may appear.

The driver consists of two parts, and there will be a dialog-window asking if you would like to install the device software for both of them.

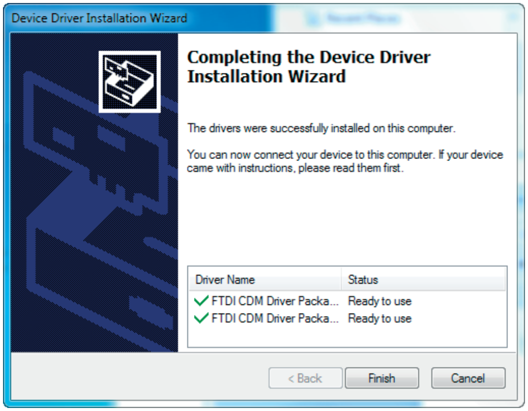
Click “Install”.

When the device driver installation is complete, the following dialog-window will appear.





Click “Finish”.



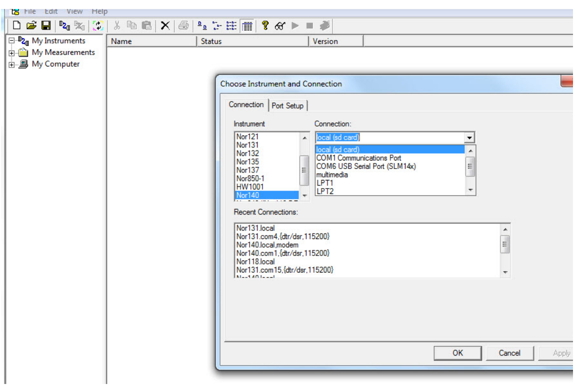
After you have plugged in the Nor13x/Nor14x for the first time, it may take some time (less than a minute) before Windows “activates” the driver.

After you have plugged in the Nor13x/Nor14x, you may check if the driver is working properly by starting Device Manager and checking the “Ports” section. Here you will also see which COM port is assigned to the attached device.

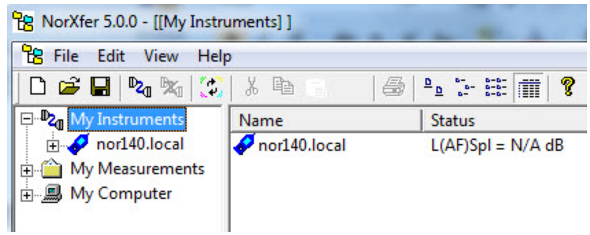
Quick Reference Guide – transfer and convert a measurement to Excel

After a series of measurements are taken and stored, the files must be transferred to the PC and converted into a readable format by the NorXfer program. The program must be installed on the PC with an appropriate USB driver installed. Follow the installation procedure given when installing the program the first time. Right click on “My Instruments” and select “connect”. The menu below appears. Select USB.

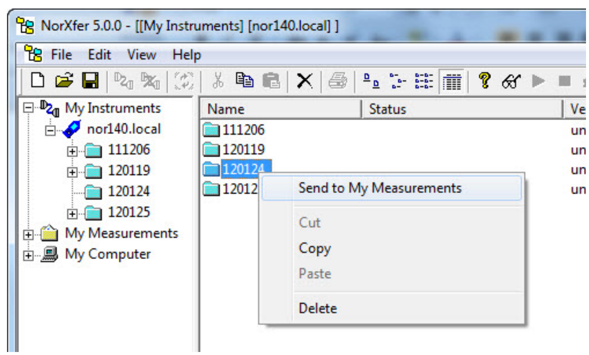
The instrument will appear as shown below. Double-click on the instrument in the right frame to expand the file folder.

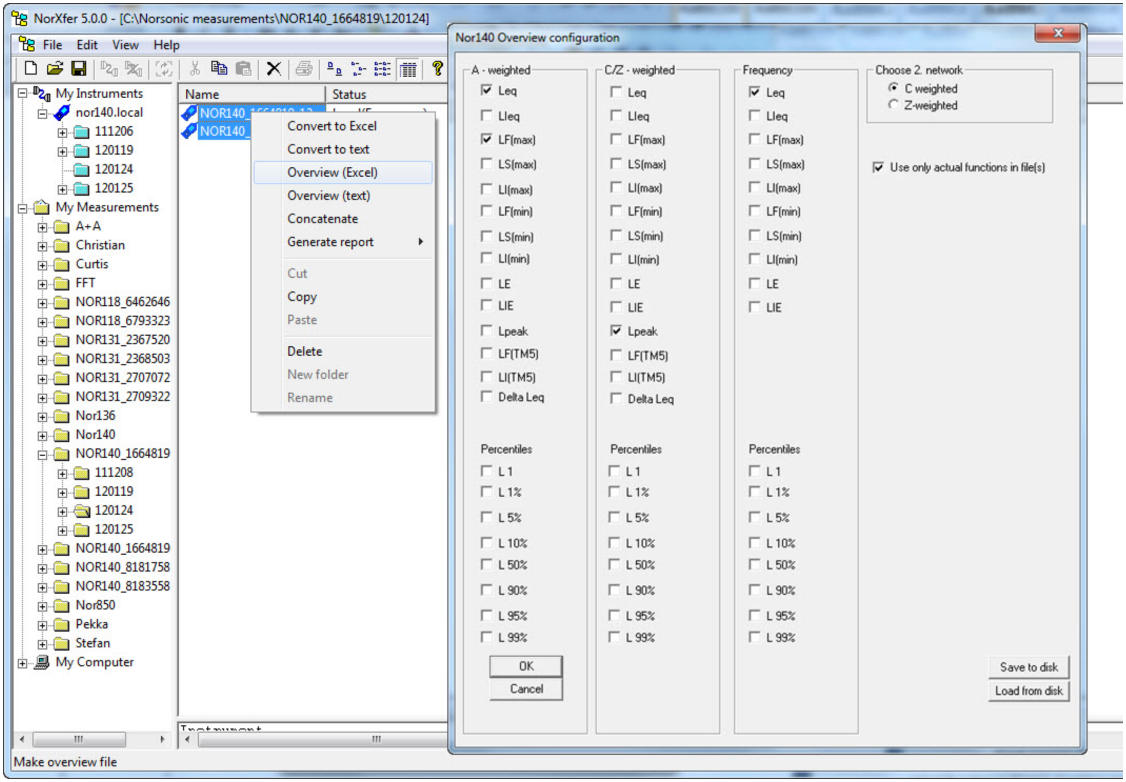


Mark the measurement files that you want to copy onto the PC. Use drag and drop to the “My measurements” folder or right click and select “Send to My Measurements”

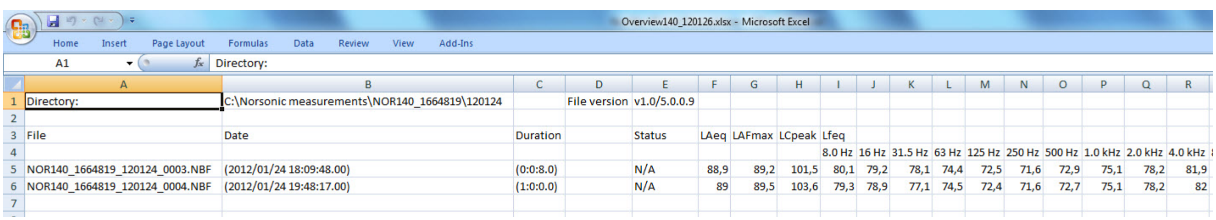


Expand “My Measurement” folder and select the right subfolder and catalogue. Select the measurement files and right click. Select “Overview (Excel)”. The Overview configuration menu appears. “Use only actual function in file(s)” checkbox and select the OK button.





An Excel file is generated that contains all the measured values. Measurements in rows, values in columns.



	A	B	C	D	E	F	G	H	I	J	K	L	M	N	O	P	Q	R
1	Directory:	C:\Norsonic measurements\NOR140_1664819\120124		File version	v1.0/5.0.0.9													
2																		
3	File	Date	Duration		Status	LAeq	LAFmax	LCpeak	Lfeq									
4									8.0 Hz	16 Hz	31.5 Hz	63 Hz	125 Hz	250 Hz	500 Hz	1.0 kHz	2.0 kHz	4.0 kHz
5	NOR140_1664819_120124_0003.NBF	(2012/01/24 18:09:48.00)	(0:0:8.0)		N/A	88,9	89,2	101,5	80,1	79,2	78,1	74,4	72,5	71,6	72,9	75,1	78,2	81,9
6	NOR140_1664819_120124_0004.NBF	(2012/01/24 19:48:17.00)	(1:0:0.0)		N/A	89	89,5	103,6	79,3	78,9	77,1	74,5	72,4	71,6	72,7	75,1	78,2	82
7																		

Virtual Instrument – Nor1036 Installation procedure

wnload the program from

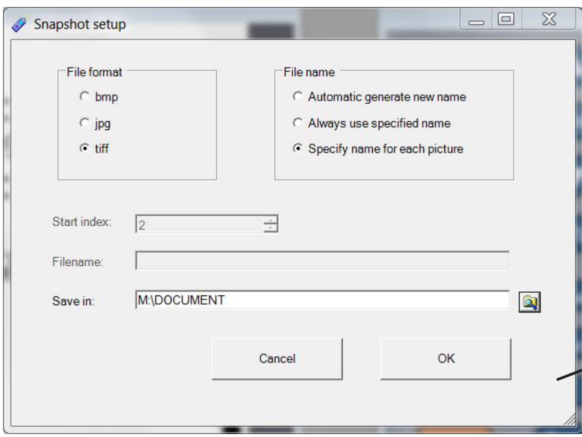
www.norsonic.com/downloads.

Follow the procedure given by the installation program.

Using Virtual Instrument Nor1036

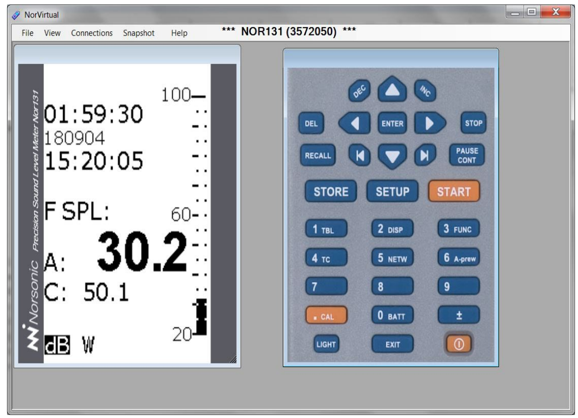
Connect the instrument to the PC using the USB interface cable Nor4525

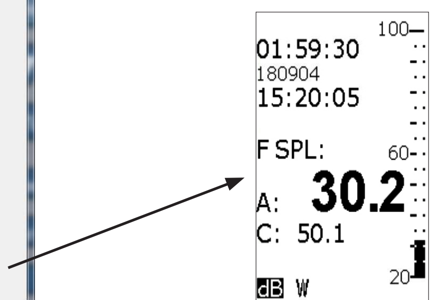
Start the program Virtual instrument Nor1036. The program will automatically start to look for the instrument. This may take some time. The screen below will be shown when the program is running. A copy of the instrument display is shown adjacent to picture of the keyboard for the instrument. You can operate the instrument as usual by pointing on the appropriate key with the PC-mouse and click with the left-hand mouse-button.



If you have more instruments connected, select the wanted unit in the menu “Connections”.

You may store a digital copy of screen by pressing “Snapshot”. The setup for the picture format and where to store the picture is found in the “File” menu. The figure below shows the setup. The screen picture may be stored as Bitmap (bmp), Jpeg or Tiff. The Tiff-format is reversed: white text on black background.





Technical specifications

Unless stated otherwise, the specifications are for the complete sound level meter Nor131/Nor132 equipped with microphone.

Nor131 is equipped with detachable preamplifier type Nor1207 and microphone Nor1228. Nor132 comes with a fixed preamplifier and microphone Nor1229.

Values are based on the nominal value for the microphone sensitivity, –26.0 dB relative to 1V/Pa, and 6 dB attenuation in the preamplifier. Each sound level meter is individually calibrated.

A microphone cable Nor4531 of length 5 m may be used between the microphone preamplifier and the instrument body without loss of performance (Nor131 only). Longer cables may be used if maximum sound pressure level or frequency is reduced.

The definition of terms is based on IEC 61672-1: 2002-05. The options included in the basic instrument may vary. Please check with your local supplier for the latest information.

Type of instrument

Nor131: Sound level meter IEC 61672-1, class 1, group X, measuring exponential time-weighted levels, integrating-averaging levels and sound exposure levels. The instrument also complies with requirements in the previous International standards for sound level meters: IEC 60651 type 1 and IEC 60804 type 1

If 1/-1 or 1/3- octave band filters are installed, the instrument complies with IEC 61260, class 1.

Nor132: Sound level meter IEC 61672-1, class 2, group X, measuring exponential time-weighted levels, integrating-averaging levels and sound exposure levels. The instrument also complies with requirements in the previous International standards for sound level meters: IEC 60651 type 2 and IEC 60804 type 2. Note that the instrument may be applied in the extended temperature range -10°C to +50°C compared to the requirements for IEC 61672-1, class 2

If 1/1- or 1/3-octave band filters are installed, the instrument complies with IEC 61260, class 1.

Analogue input

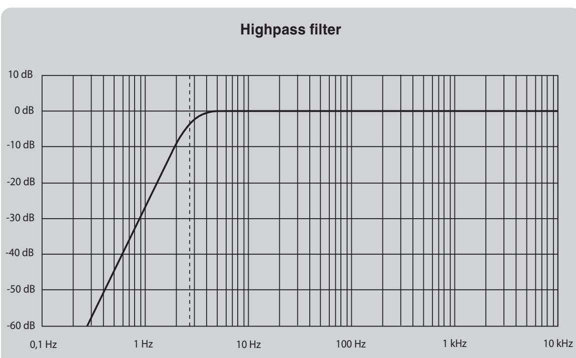
Number of channels: 1
Input connector: TNC-connector for Nor131, no connector for Nor132.
Preamplifier supply: IEPE-type, 3 mA /24 V
Polarisation voltage: 0 V (prepolarised microphone).
Maximum input signal: ± 11 V peak
Input impedance (Nor131): More than 800 kohm, less than 250 pF

Measurement range (line input): 0.3 mV to 7 V (RMS) in one range corresponding to –10 dB to 137 dB with a microphone sensitivity of 50 mV/Pa. The maximum peak value ±10 V corresponds to 140 dB. Lower limits depend on the network/filter applied.

Highpass filter

The input section is equipped with a highpass filter to reduce noise from wind or other sources with frequencies below the frequency range for measurements.

Filter type: 3rd order HP filter (-3 dB at 2.7 Hz, Butterworth response, see figure below).



Analogue to digital conversion

The analogue input signal is converted to a digital signal by a multirange sigma-delta converter with an effective sampling frequency of 48 kHz. The anti-aliasing filter is a combination of an analogue and a digital filter.

Frequency weightings

Simultaneous measurement of A- and C-weighting or A- and Z-weighting. 1/1 octave band levels may be measured simultaneously if options providing these weightings are installed.
1/1-filters: 8, 16, ...16000 Hz, class 1, digital IIR filters, base 10 system according to IEC 61260.
1/3-filters: 6.3, 8, 10, 12.5, 16, ...20000 Hz, class 1, digital IIR filters, base 10 system according to IEC 61260.

Level detector

Detector type: Digital true root-mean-square (RMS) detection and peak detection, resolution 0.1 dB.

Crest factor capability: The crest factor is only limited by the peak-value of the signal.

Overload indicator: Should the input signal exceed 10V peak, a triangle shaped overload indicator ▲ will appear on the top of the vertical bargraph in the SLM display. If this happen during a running measurement, the overload status will remain in the display as an empty triangle △ even after the signal has decreased below the 10V peak.

The overload status for the measurement is stored in the memory together with the measurement data. When a new measurement is started, the internal instrument overload status is reset automatically.

Underrange indicator: The instrument Standard IEC 61672 requires that sound levels below the lower measurement range is indicated. The Nor140 indicates this status in the normal SLM display in which the current level has the sign '<' ("less than") to the left of the value.

The <-sign appears as a display feature when the measured level on any weighting network is less than 24 dBA, 30 dBC or 40 dBZ.

Please note that the underrange status is only a display warning and it is not stored in the memory together with the measurement data.

Time weightings and measured functions

Measurement of the following functions:

- F–time-weighted sound pressure level, instantaneous
- Maximum F–time-weighted sound pressure level
- Minimum F–time-weighted sound pressure level
- S–time-weighted sound pressure level, instantaneous
- Maximum S–time-weighted sound pressure level
- Minimum S–time-weighted sound pressure level
- I–time-weighted sound pressure level, instantaneous
- Maximum I–time-weighted sound pressure level
- Minimum I–time-weighted sound pressure level
- Integrated-averaged sound pressure level
- Sound exposure level
- Peak sound level

As an option, the sound level meter may also measure:

- Integrated-averaged I–time-weighted sound pressure level
- –time-weighted sound exposure level
- Taktmaximalpegel – DIN 45657, F time response, 5 seconds Takt.

Level distribution

As an optional extension, the instrument may be fitted to calculate the exceeding level (cumulative level distribution) for the F time weighted level. The calculation is done for frequency weightings A and C or Z and for 1/1 octave band levels (if filter option is installed).

Class width: 0.2 dB
Number of classes: 652 for levels between 10 dB above full scale (140 dB) and 120 dB below full scale (10 dB). The classes for the highest and lowest levels are extended to also include levels above and below, respectively.
Sampling frequency for level: 10 samples per second
Display resolution: 0.1 dB based on interpolation

Indication range

The calibration of the instrument allows microphones with sensitivity in the range –84 dB to +15.9 dB relative to 1 volt/pascal to be applied. The corresponding display range for the indicated sound level is –50 dB to +180 dB. Preamplifier attenuation is assumed to be in the range 0 to 9,9 dB.

Self-noise levels

The self-noise is measured with the calibration set to –26.0 dB corresponding to a microphone sensitivity of 50 mV/Pa. For voltage input, the level 0 dB then corresponds to 1μV. Typical values for the self-noise are 3 to 5 dB lower than the values stated.

Dummy microphone: Noise measured with 18 pF microphone dummy and microphone preamplifier Nor1207 with a nominal attenuation of 6 dB, averaged over 30 s of measurement time:
A-weighted: 18 dB, C-weighted: 20 dB,
Z-weighted: 25 dB

Real microphone: Noise measured with Nor1228 microphone and preamplifier Nor1207 with a nominal attenuation of 6 dB, averaged over 30 s of measurement time:
A-weighted: 20 dB, C-weighted: 27 dB, Z-weighted: 35 dB.

Line input: Noise measured with the input terminal on the sound level meter short-circuited to ground, line input selected, averaged over 30 s of measurement time:
A-weighted: 8 dB, C-weighted: 10 dB,
Z-weighted: 15 dB.

Field calibration

The recommended sound calibrator for verification of the sensitivity of the sound level meter Nor131 is Norsonic Nor1251 with a nominal sound pressure 114.0 dB at 1 kHz. Recommended sound calibrator for Nor132 is Nor1251 or Nor1252 (Class 2, also with a nominal sound pressure 114.0 dB at 1 kHz). In order to compensate for effects due to diffraction around the microphone, we recommend adjusting the sound level meter to indicate 113.8 dB (random incidence correction off). If random incidence correction is on, the sound level meter shall be adjusted to 114.0 dB.

If other types of calibrators are used for the calibration, we recommend adjusting the sound level meter to indicate the following levels referred to the sound pressure acting on the diaphragm of the microphone (random incidence correction off):

Freq	**125 Hz**	**250 Hz**	**1 kHz**	**4 kHz**	**8 kHz**
Level	0.0 dB	0.0 dB	–0.2 dB	–0.8 dB	–2.8 dB

Measurement duration and resolution

The total time period for a measurement may be set from 1 second up to 100 hours less 1 second with 1 second resolution.

Total range for measurement of A-weighted levels

The linear operating range is identical to the total range.

Frequency	**31.5 Hz**	**1 kHz**	**4 kHz**	**8 kHz**	**12.5 kHz**
Upper level	98 dB	137 dB	138 dB	136 dB	133 dB
Lower level	24 dB	24 dB	24 dB	24 dB	24 dB
Ref level test	94 dB	114 dB	114 dB	114 dB	114 dB

The primary indicator range for compliance with IEC 60651 type 1 is 24 dB to 117 dB. For compliance with IEC 60804 type 1, the linearity range is 24 to 137 dB, and the pulse range 24 dB to 140 dB, respectively.

Total range for measurement of C-weighted levels

The linear operating range is identical to the total range.

Frequency	**31.5 Hz**	**1 kHz**	**4 kHz**	**8 kHz**	**12.5 kHz**
Upper level	134 dB	137 dB	136 dB	134 dB	131 dB
Lower level	30 dB	30 dB	30 dB	30 dB	30 dB
Ref level test	114 dB	114 dB	114 dB	114 dB	114 dB

Total range for measurement of Z-weighted levels

The linear operating range is identical to the total range.

Frequency	31.5 Hz	1 kHz	4 kHz	8 kHz	12.5 kHz
Upper level	137 dB	137 dB	137 dB	137 dB	137 dB
Lower level	40 dB	40 dB	40 dB	40 dB	40 dB
Ref level test	114 dB	114 dB	114 dB	114 dB	114 dB

Measurement range for C-weighted peak levels

Frequency	31.5 Hz	1 kHz	4 kHz	8 kHz	12.5 kHz
Upper level	137 dB	140 dB	139 dB	137 dB	134 dB
Lower level	45 dB	45 dB	45 dB	45 dB	45 dB
Ref level test	114 dB	114 dB	114 dB	114 dB	114 dB

Power supply

Battery: 4 cells, IEC LR6, AA-sized, Alkaline batteries are recommended (e.g. Duracell Ultra M3). AA-sized NiCd or NiMH rechargeable batteries may be used, but must be charged outside the instrument. Battery voltage and time on battery since last change of batteries are indicated.

Typical battery life time (Duracell Ultra M3)**:** 8 – 12 hours.

External dc: 11 – 16 volt. Power consumption approximately 1.2 watt dependent on selected modes of operation. External DC source should have source-impedance less than 1 ohm and be able to supply at least 300 mA. The mains adaptor Nor340 is recommended for use with the instrument.

If the external supply falls below 9 V, the instrument will use the internal batteries if available. If the instrument has switched off due to loss of power or insufficient supply voltage, the instrument will automatically switch on after reapplying the external DC supply.

Socket for external dc: 1.3 mm plug, negative voltage on centre-terminal.

The instrument will automatically switch off if the battery or external voltage is too low for operation within the stated specifications. The maximum battery voltage for conformance testing is 4 × 1.6 V = 6.4 V.

The instrument has a calendar clock supplied from the batteries or external DC-supply. The clock is supplied from a charged capacitor during change of batteries.

Display

The display is a monochrome, transreflective LCD graphical display with 160×240 pixels (W×H) with automatic temperature compensation for contrast and viewing angle. Pressing the light key illuminates the display. The light switches off automatically 2 minutes after the last operation of any key.

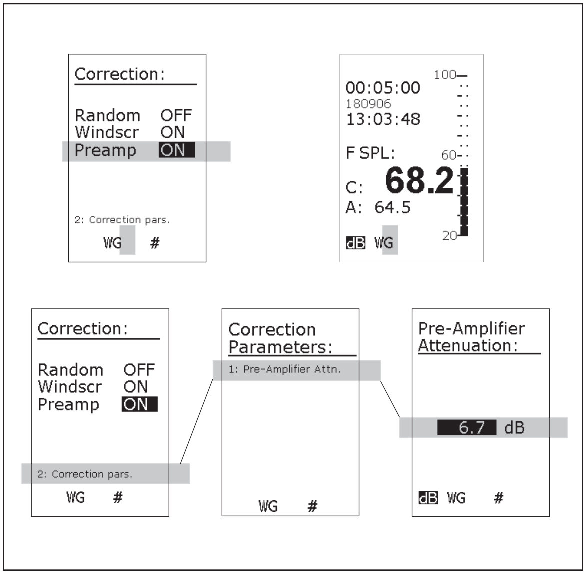
The bar graph display covers 80 dB, which may be scrolled in 10 dB steps to cover the total range.

Keyboard

The keyboard is of silicon-rubber type.

Adjustment of indicated levels

Random response. The instrument is normally equipped with a microphone with flat free-field response and satisfies the class 1 and class 2 requirements in IEC 61672-1 to free-field response for Nor131 and Nor132 respectively. By selecting the random response correction network included, the instrument will satisfy the similar class requirements in IEC 61672-1 and ANSI S1.4-1997 to random response. The nominal correction to obtain flat random response is shown in the adjacent figure.



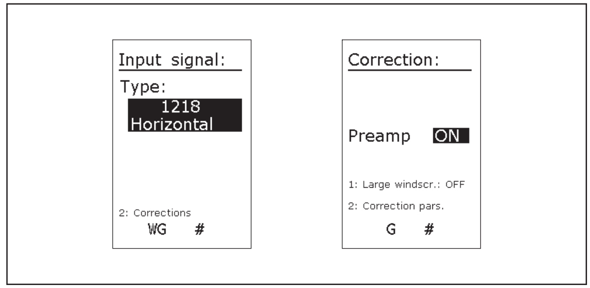
Activating random response correction:

(Not available in German version)

- Press **SETUP** > **1** (Instr.) > **4** > **2**(Correct.) to gain access to the Corrections menu. Navigate in the menu as usual and activate the correction parameter Random by means of the **INC** and **DEC** keys. Do the same to deactivate. Random response correction activated is indicated by an **R** in the lower line of the display.

Windscreen

The instrument may be used with windscreen Nor1451. The windscreen correction has to be switched on to obtain the stated specifications. The nominal correction for the windscreen correction network is shown in the figure next page.



Activating windscreen correction

- Press **SETUP** > **1** (Instr.) > **4** > **2** (Correct.) to gain access to the Corrections menu. Navigate in the menu as usual and activate the correction para-meter Windscr by means of the **INC** and **DEC** keys. Do the same to deactivate. Windscreen correction activated is indicated by a **W** in the lower line of the display.

Activating corrections for Nor1218

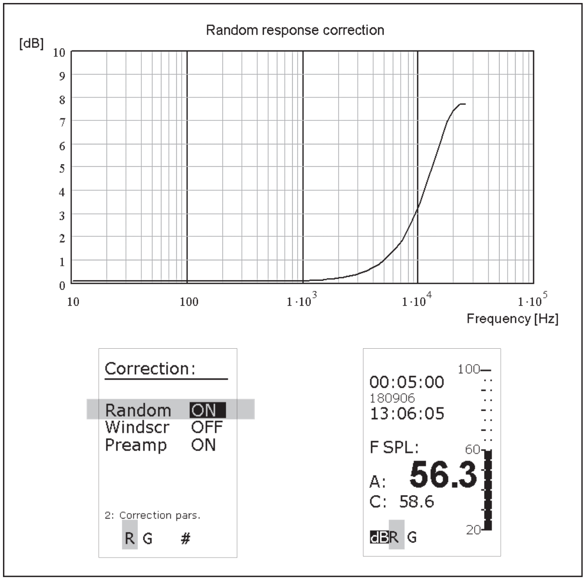
To activate a correction network for this do as follows:

- Press **SETUP** > **1** (Instr.) > **4** > Select “Nor1218 Horizontal”.

Activating corrections for large wndscreen-Nor4529

The Nor4529 large windscreen can be used with the “Nor1218 Horizontal” microphone. To activate a correction network for this do as follows:

- Press **SETUP** > **1** (Instr.) > **4** > Select “Nor1218 Horizontal” > **2** (Corrections) > **1** Preamp ON or OFF.

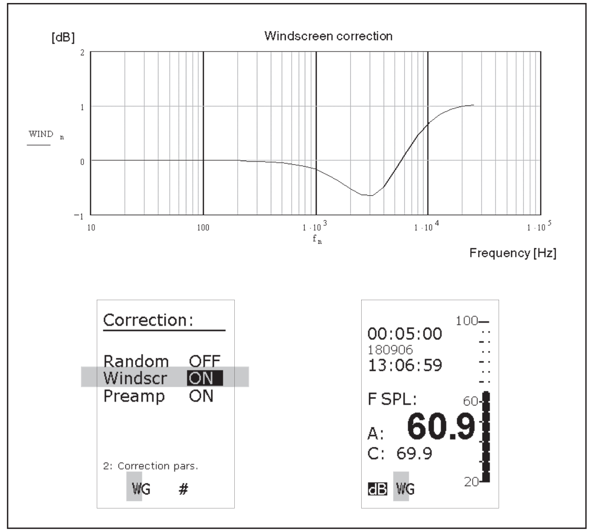


Preamplifier attenuation

The instrument has the ability to correct for the attenuation in the preamplifier. Typical value of the attenuation is 6 dB. The correction can be set in the range 0.0 to 9.9 dB. The correction can be switched on/off when microphone is selected as the source. When Line is selected, the preamplifier correction is automatically switched off.

Activating the preamplifier attenuation:

- To activate the preamplifier attenuation press **SETUP** > **1** (Instr.) > **4** > **2** (Correct.) to gain access to the Corrections menu. Navigate in the menu as usual and activate the correction parameter Preamp by means of the **INC** and **DEC** keys. Do the same to deactivate. Preamplifier attenuation activated



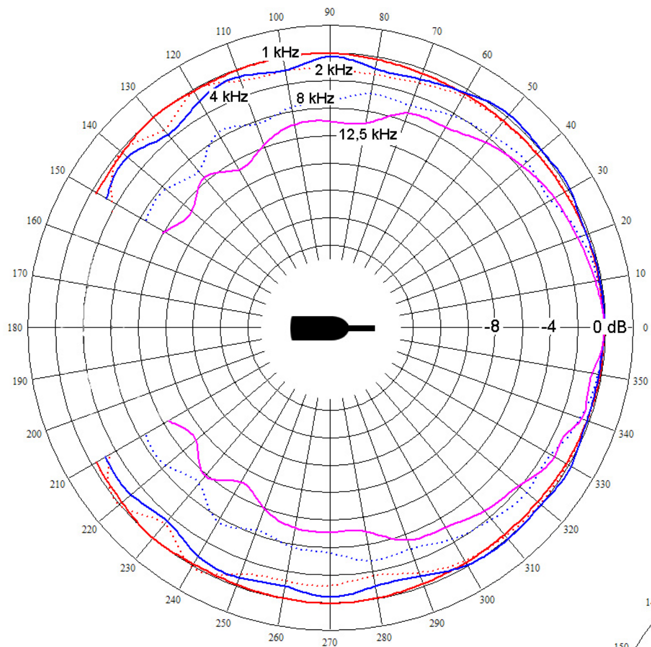
is indicated by a **G** (for Gain) in the lower line of the display. (In the German version of the program correction is not accessible from the user interface.)

Setting the amount of attenuation:

- In the Correction menu, press **2** (Corr.par) to gain access to the correction parameter setup menu. Press **1** to gain access to the attenuation setting itself. Never change this setting unless you know what you're doing! Use the numerical keypad to set the attenuation value. Press enter twice to leave the menu. See Fig. on the previous page for menu details

Diffraction around the instrument casing

The instrument casing is designed to have low effects on the sound measured at the microphone. The figures on following pages shows the measured effect of the instrument casing at reference environmental conditions.

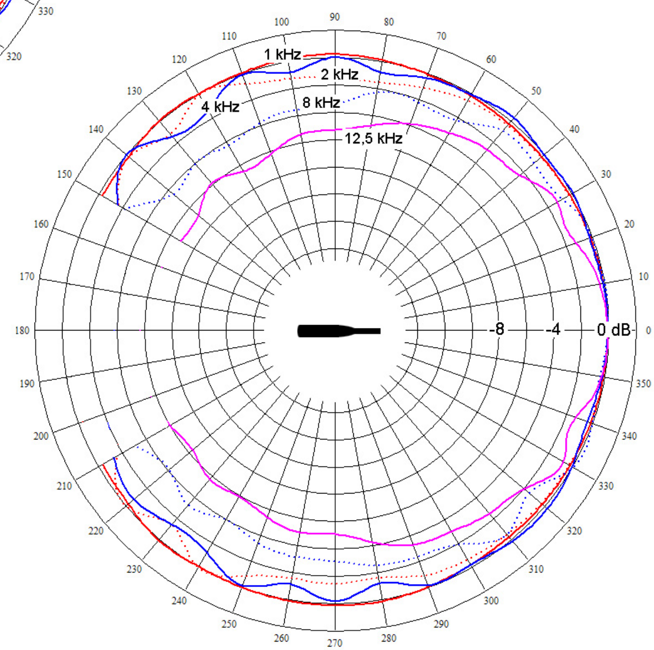


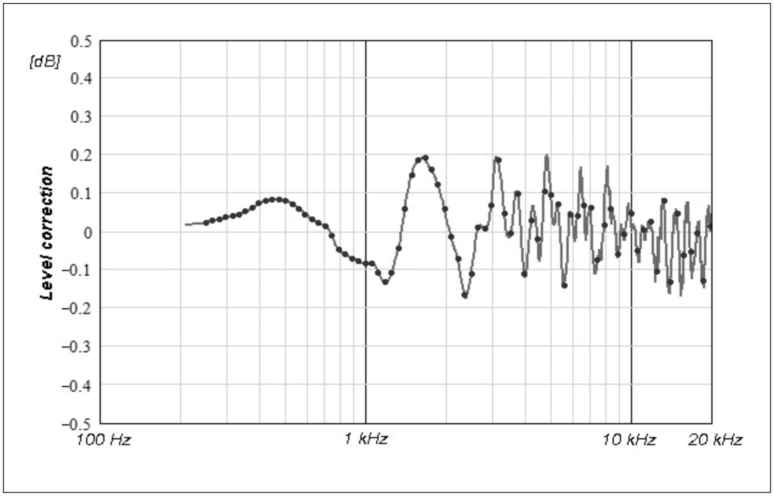
Nor131 (Hor)

Nor131 directional response, measured in the horizontal plane through the centre of the microphone. 0 ° corresponds to sound approaching the microphone from the front along the axis of symmetry. The instrument display faces upwards.

Nor131 (Vert)

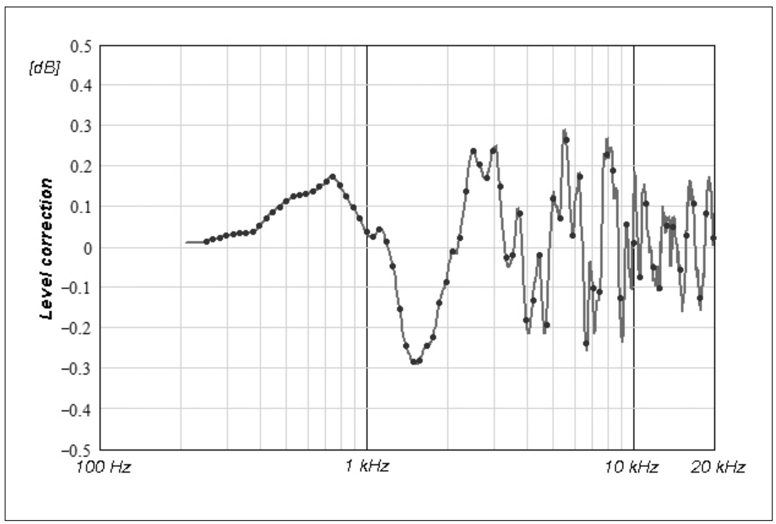
Nor131 directional response, measured in the vertical plane through the centre of the microphone. 0 ° corresponds to sound approaching the microphone from the front along the axis of symmetry. The instrument display faces upwards.





Case reflections for Nor 131 for sound approaching the microphone from the front along the axis of symmetry. The diagram shows the level difference between the response for the microphone-preamplifier alone (extension cable) and when mounted on the instrument

Case reflections for Nor 132 for sound approaching the microphone from the front along the axis of symmetry. The diagram shows the level difference between the response for the microphone-preamplifier alone and when mounted on the instrument.



Analogue output

The analogue output is a reproduction of the digitized input signal from the microphone (or input connector) obtained by a digital to analogue converter.
Output voltage: Full scale on the display corresponds to 100 mV.
Output impedance: The loading impedance shall be 16 ohm or greater. The output is short-circuit proof to GND.
Gain accuracy at 1 kHz: ±0.2 dB.
Frequency response re. 1 kHz:
±0.5 dB for 20 Hz < f < 16 kHz.

USB port

USB type1.1

Data storage

Measured data is stored in the internal memory of the sound level meter. The memory is of the "flash" type retaining the information without battery supply. Approximately 5 Mbyte is available for the data storage. This corresponds 10000 measurement for sound level meters without octave analysis and up to 2500 for sound level meters with octave analysis.

Environmental conditions

Reference conditions. The reference conditions for the instrument are as specified by IEC 61672-1
Temperature: 23°C
Humidity: 50% RH
Atmospheric pressure: 101.325 kPa

Environmental condition for operation

Temperature: –10°C to +50°C
Humidity: 5% to 90% RH, dewpoint less than 40°C
Atmospheric pressure: 85 kPa to 108 kPa

Environmental condition for storage

Temperature: –30°C to +60°C
Humidity: 5% to 90% RH, dewpoint less than 40°C
Atmospheric pressure: 50 kPa to 108 kPa

Warm-up time

The warm-up time for the main instrument is very short and the instrument obtains the stated accuracy as soon as the self-test is made (20 sec). Before a re-calibration is attempted, at least two minutes for warm-up is recommended.

Sensitivity for vibration

If the instrument is used under strong vibrational conditions, it is recommended to use an extension cable between the preamplifier and the instrument body (Nor131). The vibration will mainly affect the microphone, which is most sensitive if the vibration is applied perpendicular to the diaphragm. Typical values are 55 dB to 65 dB for acceleration values of 1 ms-2 perpendicular to the diaphragm.

Sensitivity for magnetic fields

The maximum indication for exposure to magnetic field of 80 A/m and any orientation is typically less than 20 dB.

Size and weight

Depth: 29 mm
Width: 74 mm
Length, excl. microphone/preamplifier: 215 mm
Length, incl. microphone/preamplifier: 305 mm
Weight incl. batteries: 380 g

Information for conformance testing

Reference Sound Pressure Level: 114.0 dB re 20 μPa. The reference frequency is 1000 Hz.

Reference Level Range: The instrument has one level range only.

Microphone Reference Point and Direction: The microphone reference point is the geometric centre of the diaphragm of the microphone. The microphone reference direction is from the microphone and along the axis of rotational symmetry for the microphone and preamplifier.

Battery voltage: The instrument will automatically switch off if the battery or external voltage is too low for operation within the stated specifications. The max. battery voltage for conformance testing is 4 ×1.6 V = 6.4 V.

Electromagnetic Compatibility: When the instrument is tested for conformance to electromagnetic compatibility requirements, the instrument should be in the measurement mode, as this normally will generate the highest levels of emissions. The highest susceptibility is normally observed when the display faces the principal direction of propagation for the electromagnetic field.

SIAFA
www.siafa.com.ar

Norsonic

Declaration of Conformity

We, NORSONIC AS, GUNNERSBRÅTAN 2, N-3409 TRANBY, NORWAY, declare under our sole responsibility that the product:

Sound Level Meter / Real Time Analyser Nor131 and Nor132

to which this declaration relates, is in conformity with the following standards or other normative documents

Standards:	IEC61672-1 CLASS 1[1] OR 2[2]	ANSI S 1.4 1983 TYPE 1[1] OR 2[2]
	IEC 60651 TYPE 1[1] OR 2[2]	ANSI S 1.43 1997 CLASS 1[1] OR 2[2]
	IEC 60804 TYPE 1[1] OR 2[2]	ANSI S1.11-2004 CLASS1[1] OR 2[2]
	IEC 61260 CLASS 1	EN 61010-1: FEBRUARY 2001

following the provisions of the EMC-DIRECTIVE.

Note: [1]: Nor131
[2]: Nor132

This product has been manufactured in compliance with the provisions of the relevant internal Norsonic production standards. All our products are tested individually before they leave the factory. Calibrated equipment – traceable to national and international standards – has been used to carry out these tests.

During the RF emission test the following was connected: USB cable (1m), mains adapter Nor340, microphone preamplifier Nor1207 and microphone Nor1228. Setup: Measurement duration 1h, Frequency mode parallel; 1/1 octave, 2nd network Z.

During the RF immunity test the following was connected: USB cable (1m), microphone preamplifier, Nor1207 and microphone Nor1228. Setup: Frequency mode parallel; 1/1 octave, 2nd network Z. Orientation: Laying face up on the table and the microphone was pointing towards the antenna.

During the AC power frequency field test the following was connected: microphone preamplifier Nor1207 and microphone Nor1228. Setup: Frequency mode parallel; 1/1 octave, 2nd network Z.

The orientation of the instrument in the magnetic field had no influence. During the ESD test the SPL value may show some fluctuations from the ESD pulse. Power supply: Battery voltage 4-6.4V. External DC voltage 11-16V.

This Declaration of Conformity does not affect our warranty obligations.

Tranby, October 2018

Jens Petter Ringvold
Jens Petter Ringvold
Cheif Engineerr

The declaration of conformity is given according to EN 45014 and ISO/IEC GUIDE 22.

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Norsonic AS supplies a complete range of instrumentation for acoustics – from sound calibrators, microphones and preamplifiers; via small handheld sound level meters to advanced, yet portable, real time analysers, but also building acoustics analysers and complete community, industry and airport noise monitoring systems. Contact your local representative or the factory for information on our complete range of instrumentation.