TECHNICAL NOTES

Precision Integrating Sound Level Meter

NL-18

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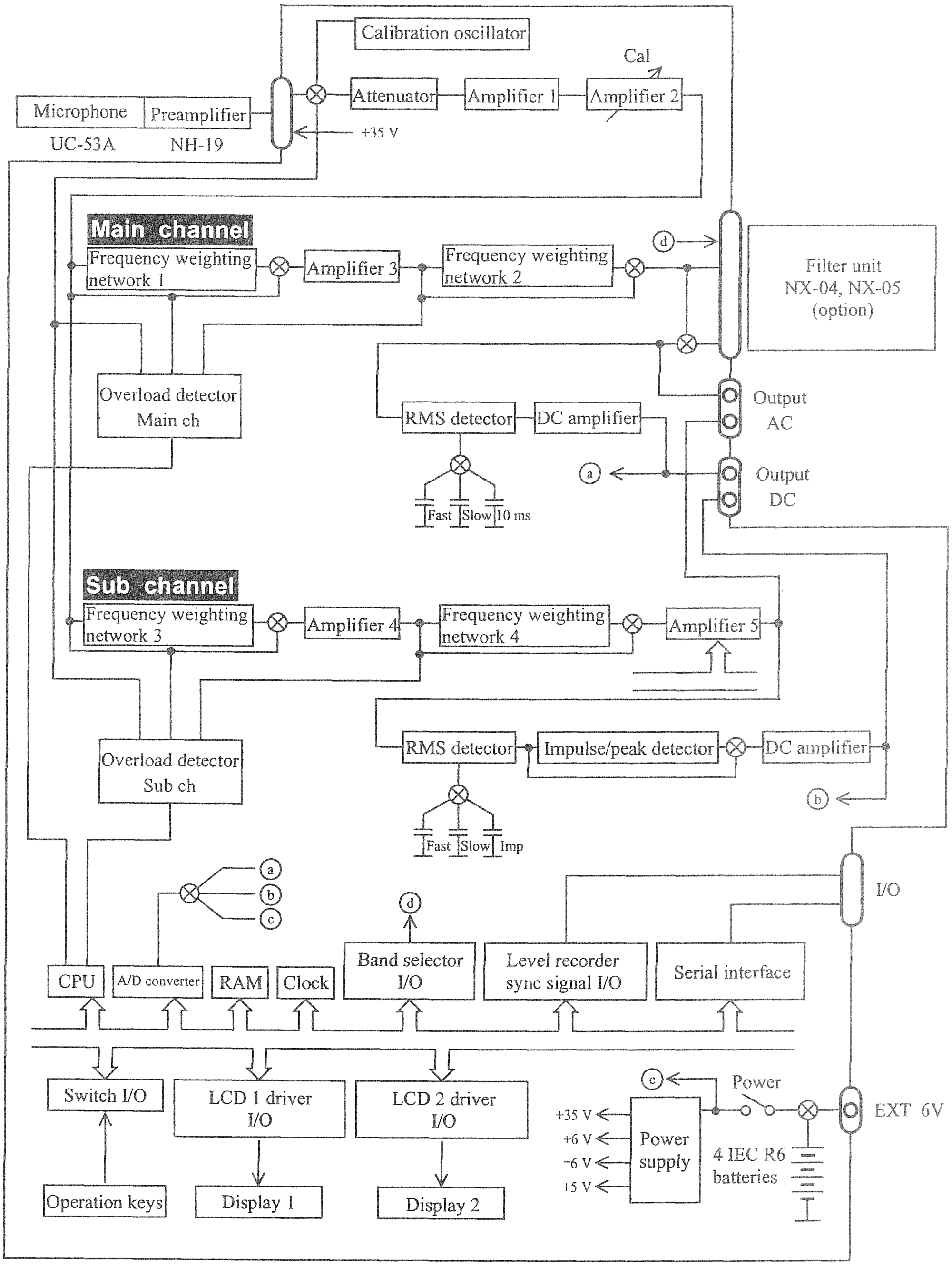
Contents

Configuration of the NL-18

A block diagram of the NL-18 is shown below, followed by a brief explanation of the various sections.



Block Diagram

Microphone preamplifier

The unit employs a 1/2-inch prepolarized condenser microphone.

Microphone: UC-53A

Preamplifier: NH-19

Calibration oscillator

This circuit supplies the 1000-Hz sine-wave signal used for electrical calibration.

Attenuator

Attenuates the signal from the preamplifier according to the level range setting.

Amplifier

Signal amplification is switched according to the level range setting. Amplifier 2 serves for calibration. Amplifier 5 is controlled by the CPU to ensure that main channel gain and sub channel gain are equal.

Frequency weighting network

This network contains filters which provide type A weighting, type C weighting, or flat.

Overload detector

Monitors the signal waveform at various circuit points.

RMS detector

True rms detection and logarithmic compression are performed on the AC signal, resulting in the level- converted DC signal. The following time constant settings (time weightings) are available.

Main channel:	Fast, Slow, 10 ms
Sub channel:	Fast, Slow, 10 ms, Imp (impulse), Peak (peak hold)

DC amplifier

Adjusts the level before A/D conversion.

Impulse/peak detector

Serves for impulse and peak hold detection (in sub channel only).

CPU, A/D converter, RAM

The level-converted DC signal is converted into a digital signal by the A/D converter. The CPU and RAM are used for carrying out the various processing functions.

Clock

Provides date and time information.

Band selector I/O

Controls the center frequency setting when the filter unit is used.

Level recorder sync signal I/O

Performs control functions when synchronized frequency analysis with the filter unit and a level recorder is carried out.

Serial interface

Controls the RS-232C interface for data exchange with a computer.

LCD driver, LCD

Display 1 shows the noise level, measurement parameters, and processing results. Display 2 shows the noise level changes over time, and is also used for showing the menu screens for parameter setup and the frequency analysis graph.

Power supply

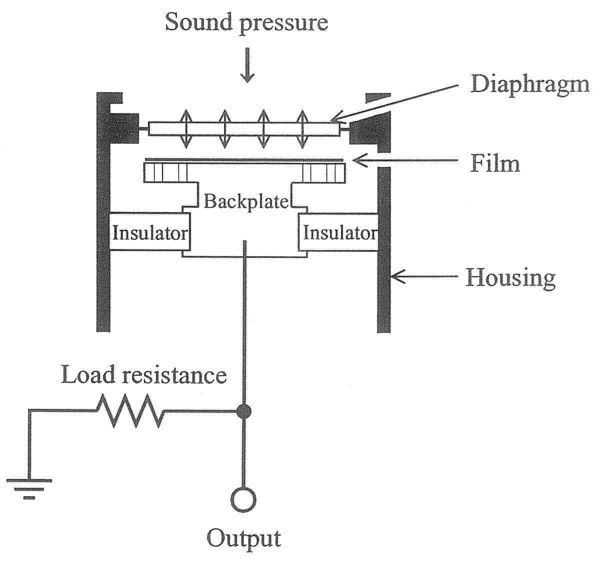
The voltage from the batteries or the AC adapter is fed to a DC-DC converter which provides the voltages required by the various sections.

Microphone

Measurements of sound pressure level can be carried out with a variety of microphone types. The precision integrating sound level meter NL-18 employs the prepolarized condenser microphone UC-53A that is compact and delivers stable and reliable response.

Construction and Operation Principle

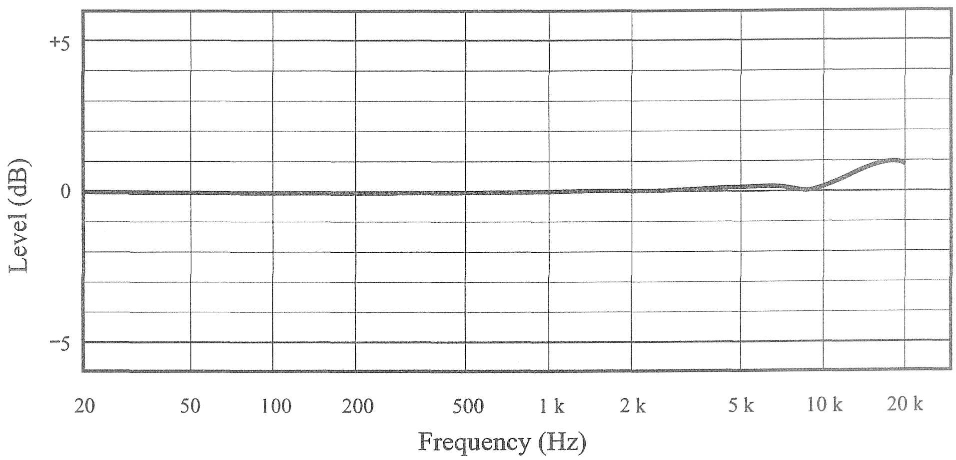
As shown in the drawing below, an electret condenser microphone consists of four main parts: diaphragm, backplate, insulator, and housing. The surface of the backplate is covered by a film holding an electrical charge. When sound pressure is applied to the diaphragm, the distance between the diaphragm and the backplate changes, thereby altering the capacitance. Using a load resistor, this change can be turned into a voltage change. The frequency response as well as the temperature and humidity characteristics of an prepolarized condenser microphone depend considerably on the type and properties of the materials used. The frequency range is determined by the resonance frequency of the diaphragm assembly.



Construction of prepolarized condenser microphone

Frequency Response

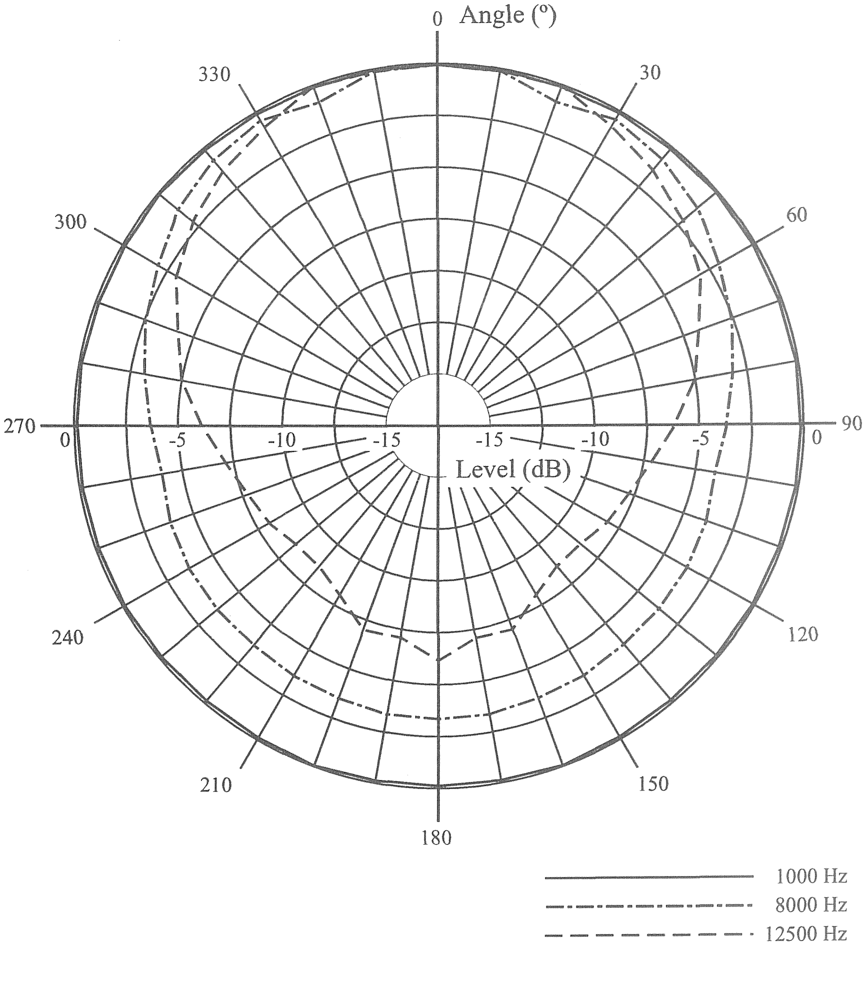
The frequency response of a sound field microphone is expressed as the frequency response in the reference direction of incidence (0°). The diagram below shows an example for the frequency response of the microphone UC-53A.



Frequency response sample of microphone UC-53A

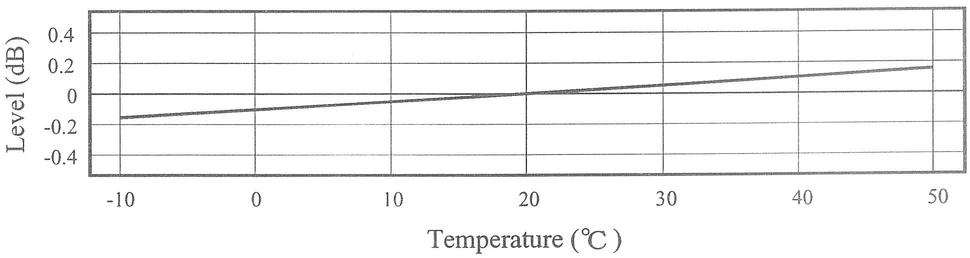
Directional characteristics

The directional characteristics of a microphone is a measure of its differing sensitivity for sound waves arriving from various angles. Since the prepolarized condenser microphone used in the NL-18 is a pressure-sensitive type, it should be equally sensitive in all directions. However, refraction and cavity effects cause a certain microphone directional characteristics at high frequencies. The diagram below shows the directional characteristics of the microphone UC-53A.



Thermal Characteristics

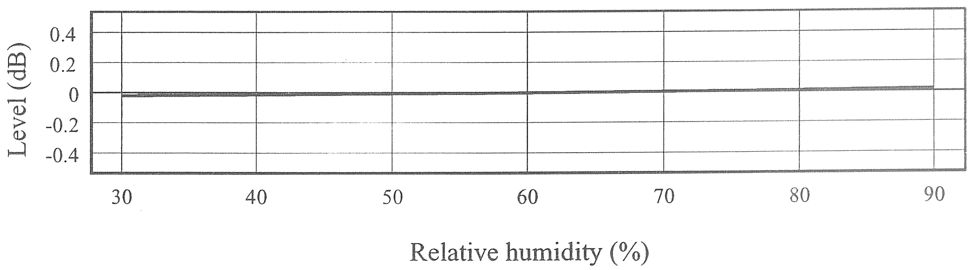
The thermal characteristics of a microphone indicate how sensitivity changes at various temperatures. This is influenced by the choice of materials and the design of the microphone. Normally, materials with a linear expansion coefficient are used. The diagrams below show the thermal characteristics of the microphone UC-53A.



Thermal characteristics (at 250 Hz)

Humidity Characteristics

The humidity characteristics of a microphone indicate how sensitivity changes at various humidity levels. The diagrams below show the microphone UC-53A.



Humidity characteristics (at 250 Hz)

Microphone Specifications

Model:	UC-53A
Nominal diameter:	1/2 inch
Sensitivity:	-28 dB (0 dB = 1 V/Pa)
Frequency response:	10 - 12500 Hz
Capacitance:	12 pF
Diaphragm type:	Titan alloy film
Temperature coefficient:	+0.005 dB/℃ (at 250 Hz)
Humidity-dependent sensitivity change:	0.1 dB or less (at 250 Hz, RH below 95%, no condensation)
Dimensions:	13.2 dia. × 12 mm

Preamplifier

Preamplifier Requirement

Since the condenser microphone is a small-capacity transducer, it has high impedance, especially at low frequencies. Therefore a very high load resistance is required to ensure uniform response extending to the low frequency range. The relationship between the microphone capacitance and the low-range cutoff frequency can be expressed as follows.

$$f_0 = \frac{1}{2\pi \cdot Z_{in} \cdot C_m}$$

f_0: Low-range cutoff frequency (Hz)
Z_{in}: Preamplifier input impedance (Ω)
C_m: Capacitance of condenser microphone (F)

If the output of the microphone were directly routed through a long shielded cable, the capacitance between the cable conductors would cause a sharp drop in sensitivity, as is evident from the following equation.

$$M_0 = \frac{C_m}{C_m + C_c} \cdot M_s$$

M_0: Output voltage into directly connected shielded cable (V)
M_s: Output voltage in microphone open condition (V)
C_c: Cable capacitance of shielded cable (F)

For the above reasons, a preamplifier is connected directly after the microphone, to provide a low-impedance output signal.

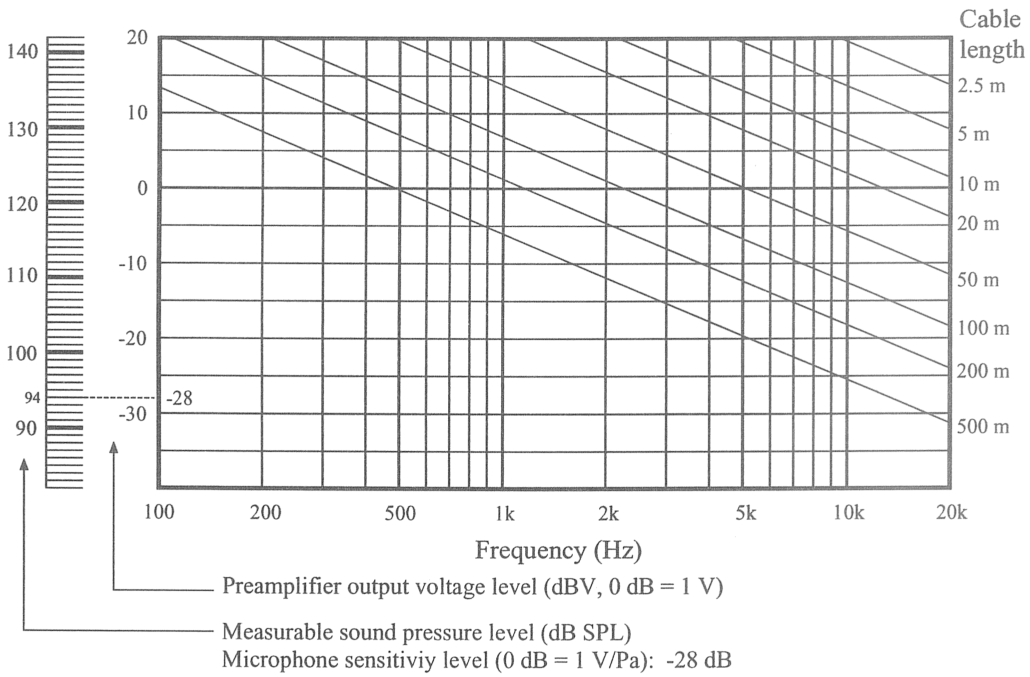
Preamplifier Specifications

Input impedance:	3 GΩ
Output impedance:	Less than 300 Ω
Maximum output current:	2 mA

Influence of Microphone Extension Cable

When the output of the microphone/preamplifier is routed through an extension cable, certain limitations regarding measurable sound pressure level and frequency range will apply. This is due to the influence of the cable capacitance. The longer the cable, the lower the measurable sound pressure level and the lower the frequency limit. The diagram below shows the relationship between cable length, measurable sound pressure level, and frequency.

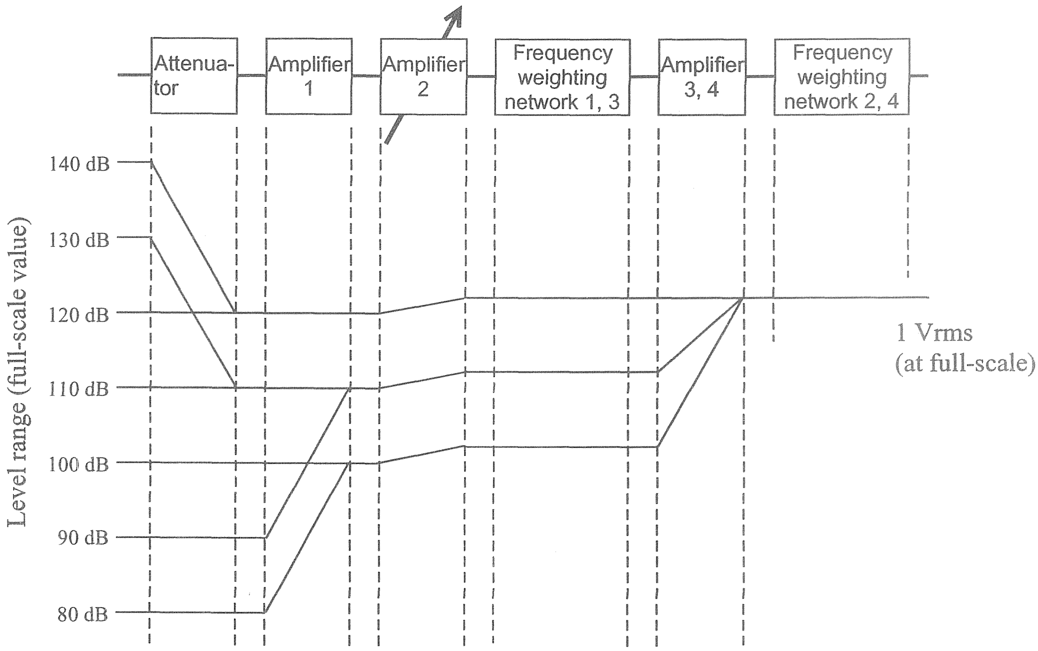
Model	Length	Model	Length
EC-04	2 m	EC-04C	30 m
EC-04A	5 m	EC-04D	50 m
EC-04B	10 m	EC-04E	100 m



If for example a sound pressure level of 120 dB is to be measured up to 3 kHz, an extension cable length of up to 100 meters can be used.

Amplifier Circuit Configuration

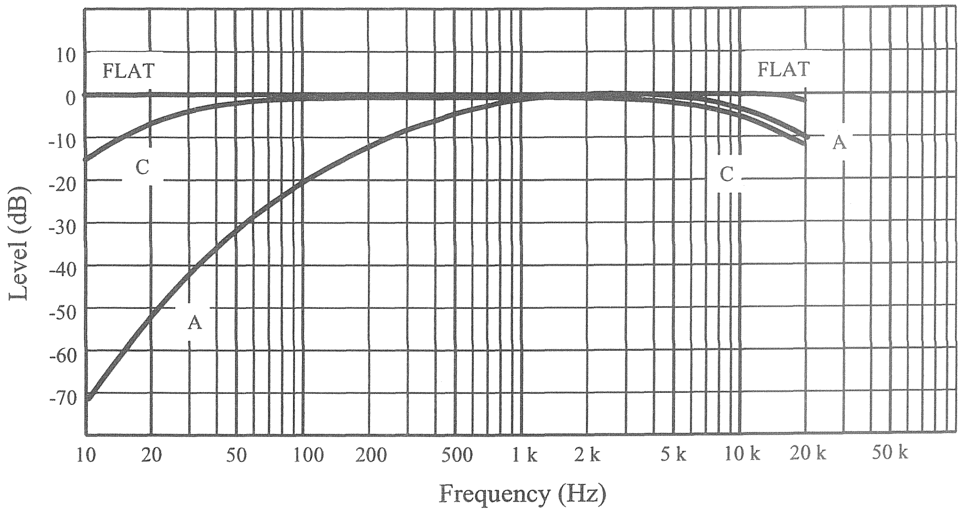
The amplifier circuit configuration and level diagram of the NL-18 are shown below. The degree of attenuation and degree of amplification depends on the level range setting. The 70 dB level range is available only when the filter unit NX-04/NX-05 is used.



Level diagram

Frequency Weighting Network

The NL-18 provides a choice between "A" and "C" weighting and flat frequency response. The electrical characteristics of the weighting network are as shown below.



Frequency weighting characteristics

The volume impression (loudness) of a sound depends not only on the sound pressure level, but also on the frequency. At high or low frequencies, a sound is felt to be less loud than a sound of equal level in the midrange. The "A" weighting curve compensates for this effect and produces measurement results which are close to the actual impression of loudness. For this reason, this type of frequency weighting is widely used for purposes such as sound level evaluation.

With the "Flat" characteristic, frequency response is linear, which is suitable for sound pressure level measurements and for using the sound level meter output for frequency analysis.

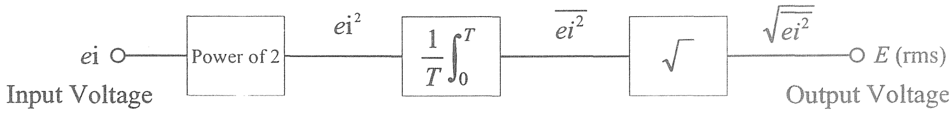
The "C" weighting curve produces almost flat response, but with a rolloff below 31.5 Hz and above 8 kHz. This is suitable for sound pressure level measurements in situations with unwanted low-frequency or high-frequency components.

RMS Detection Circuit

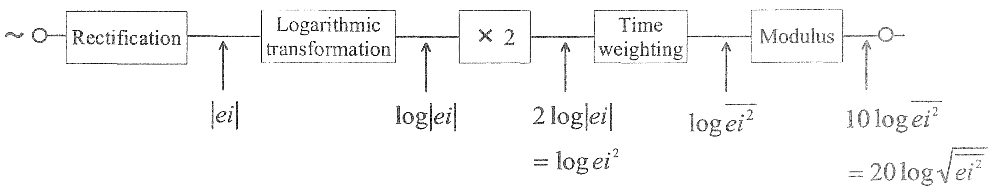
The sound level meter uses rms detection. The effective value E (rms) is defined by the following equation.

$$E(rms) = \sqrt{\frac{1}{T}\int_0^T e^2 dt}$$

The voltage e which changes over time is raised to the power of 2, and integration for the time interval T is performed. The result is divided by T and the square root is extracted. The circuit configuration for performing the above mathematical operation looks as follows.



The rms detection circuit in the NL-18 employs logarithmic transformation and is configured as follows.



For measurement and evaluation of the rms detection circuit, a signal with known crest factor is used. The crest factor is defined as crest value divided by effective value. If the signal type is known, the crest factor can be calculated precisely. The rms detection error of the NL-18 is about ± 0.2 dB for an input signal with a crest factor of 3.

During sound level measurements, the level often fluctuates drastically, which would make it difficult to evaluate readings if some kind of averaging is not applied. Sound level meters therefore provide the capability for index weighting (index averaging) using the rms circuit. The parameters of this weighting process are called the time weightings, determined by the time constant (see next page).

Sound level meters usually have a “Fast” and “Slow” setting for the time weighting. The time range that is considered for averaging is narrow in the “Fast” setting and wide in the “Slow” setting. In the “Fast” setting, the instantaneous level has a larger bearing on the displayed value than in the “Slow” setting. From the point of view of the measurement objective, the “Fast” setting is more suitable to situations with swiftly changing sound level, whereas the “Slow” setting yields a more broadly averaged picture. The “Fast” setting is more commonly used, and sound pressure level values given without other indication are usually made with “Fast” characteristics.

The “Slow” time weighting setting is suitable for measuring the average of sound with fairly constant levels. For example, in Japan aircraft noise and high-speed train noise is usually transient noise with high fluctuation, but here in Japan, the “Slow” setting is used to determine the maximum level for each noise event.

The “10 ms” setting of the NL-18 results in a very short time constant, enabling the meter to closely follow noise fluctuations.

The “Imp (Impulse)” setting enables the meter to track noise bursts of very short duration.

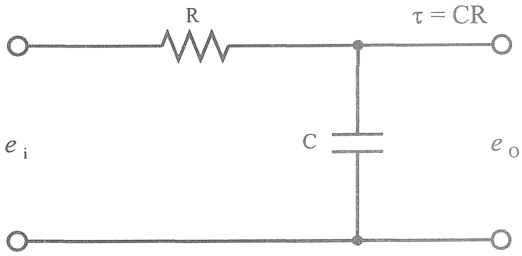
In the “Peak (Peak Hold)” condition, no averaging is carried out, and the peak value of the frequency-weighted sound pressure waveform is displayed.

Time weightings and time constant

Time weightings	Time constant	
	Rise time	Decay time
Fast	125 ms	125 ms
Slow	1 s	1 s
10 ms	10 ms	10 ms
Imp	35 ms	1.5 s

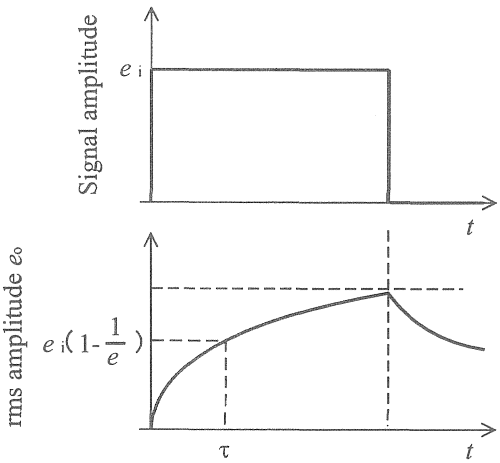
The time weighting network of the sound level meter performs index averaging on the square of the sound pressure signal. The equivalent circuit is shown at right. τ is the time constant, which equals CR.

e_i : Input voltage
e_O : Output voltage



Equivalent electrical circuit

The response of the index averaging circuit to a single burst signal is shown below.



Burst signal response

e_i : Input voltage (proportional to square of sound pressure)
e_O : Output voltage
e : Logarithm base
τ : Time constant
t : Time

Measurement Functions

L_{Aeq} (equivalent sound pressure level)

For a sound pressure level signal that changes over time, the L_{Aeq} (equivalent sound pressure level) is a hypothetical constant sound pressure level that has the same energy as the actually measured signal in the measurement interval. It is determined by the following equation.

$$L_{Aeq} = 10\log_{10} \frac{1}{Tm}\int_{t1}^{t2} \frac{pA^2(t)}{p0^2}dt$$

t_1:	Measurement start time
t_2:	Measurement end time
Tm:	Measurement time (integrated time) $Tm = t_2 - t_1$
$p0$:	Reference sound pressure 20 µPa (2×10^{-5} N / m^2)
$pA(t)$:	Instantaneous sound pressure measured with sound pressure level meter A weighting

Expressing the above equation for sound pressure level yields the following equation.

$$L_{Aeq} = 10\log_{10} \frac{1}{Tm}\int_{t1}^{t2} 10^{LA(t)/10}dt$$

$L_{A(t)}$: Instantaneous sound pressure level

In the sound pressure level meter NL-18, this statement is used as reference, and digital processing to determine L_{Aeq} is carried out according to the following equation.

$$L_{Aeq} = 10\log_{10} \frac{1}{N}\sum_{i=1}^{N} 10^{LA(i)/10}$$

N: Number of samples

Using the output signal of the rms detection circuit, digital processing is performed to determine the L_{Aeq} value. For this purpose, a suitable rms detection time weighting and sampling interval for L_{Aeq} processing must be chosen. In the NL-18, the sampling interval for A/D conversion is 10 ms (100 samples per second), and L_{Aeq} processing is carried out every interval. The L_{Aeq} reading can therefore be displayed already during measurement.

L_{AE} (sound exposure level)

The L_{AE} (sound exposure level) is a hypothetical constant 1-second sound pressure level having the same energy as a single-event sound pressure level measured with A weighting. It is determined by the following equation.

$$L_{AE} = 10\log_{10}\frac{1}{To}\int_{t1}^{t2}\frac{p_A{}^2(t)}{p_0{}^2}dt$$

t_1:	Measurement start time
t_2:	Measurement end time
To:	Reference time (1 second)
p_0:	Reference sound pressure 20 μPa (2×10^{-5} N / m^2)
$p_A(t)$:	Instantaneous sound pressure measured with sound pressure level meter A weighting

Expressing the above equation for sound pressure level yields the following.

$$L_{AE} = 10\log_{10}\frac{1}{To}\int_{t1}^{t2}10^{L_A(t)/10}dt$$

$L_{A(t)}$: Instantaneous sound pressure level

In the precision integraging sound level meter NL-18, this statement is used as reference, and digital processing is carried out according to the following equation.

$$L_{AE} = 10\log_{10}\frac{1}{N_0}\sum_{i=1}^{N}10^{L_A(i)/10}$$

N_0: Number of samples per second

In the NL-18, the sampling interval for A/D conversion is 10 ms (100 samples per second), and L_{AE} processing is carried out every interval. The L_{AE} reading can therefore be displayed already during measurement.

L_x (percentile sound pressure level)

The L_x (percentile sound pressure level) is the sound pressure level which was exceeded for x percent of the measurement time. The NL-18 allows the user to select five values for x (from 1 to 99, in 1-percent steps), and calculates the time percentile level for these five values simultaneously. The sampling interval for L_x processing is 100 ms (10 samples per second), and processing is carried out after the measurement is completed. L_x readings shown during measurement therefore are not meaningful.

L_{max}, L_{min} (maximum and minimum sound pressure level)

L_{max} is the maximum sound pressure level and L_{min} the minimum sound pressure level encountered during a measurement. In the NL-18, the sampling interval for A/D conversion is 10 ms (100 samples per second), and the L_{max} and L_{min} values since the start of the measurement are stored. Therefore the L_{max} and L_{min} readings up to the current point can be displayed already during measurement.

L_{tm3}, L_{tm5} (Takt-max sound pressure level) *

For the duration of the measurement, the maximum level within a 3-second or 5-second interval is sampled and the power average is determined. L_{tm} is calculated according to the following equation.

$$L_{tm} = 10 \log_{10} \frac{1}{N} \sum_{i=1}^{N} 10^{Lm/10}$$

L_m: Maximum level within interval (3 or 5 seconds)

N: Number of samples

The number of samples is determined according to the following equation.

For L_{tm3}: $N = \dfrac{(t_2 - t_1)}{3}$

For L_{tm5}: $N = \dfrac{(t_2 - t_1)}{5}$

t_1: Measurement start time

t_2: Measurement end time

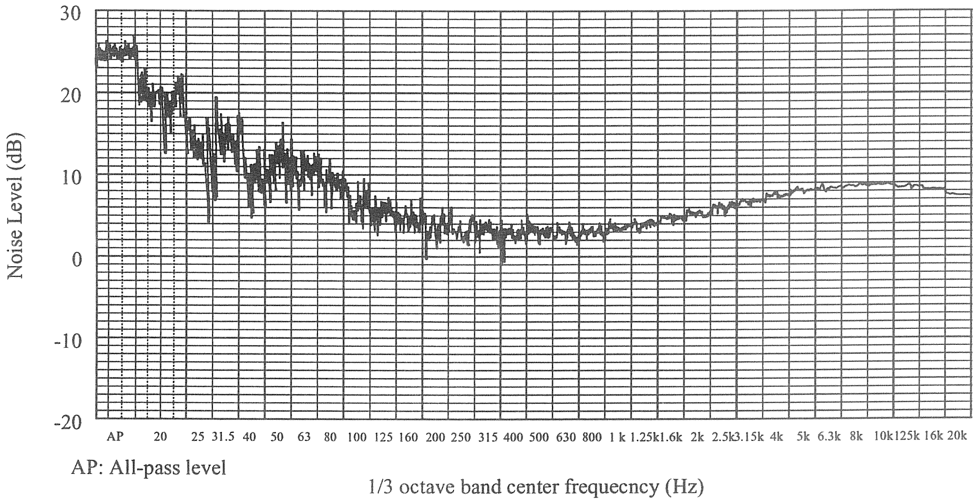
* L_{tm} is specified by DIN as "Taktmaximalpegal Mittelwert", and its meaning in English is "Power averaged maximum sound pressure level in a measuring period."

L_{peak} (waveform peak hold)

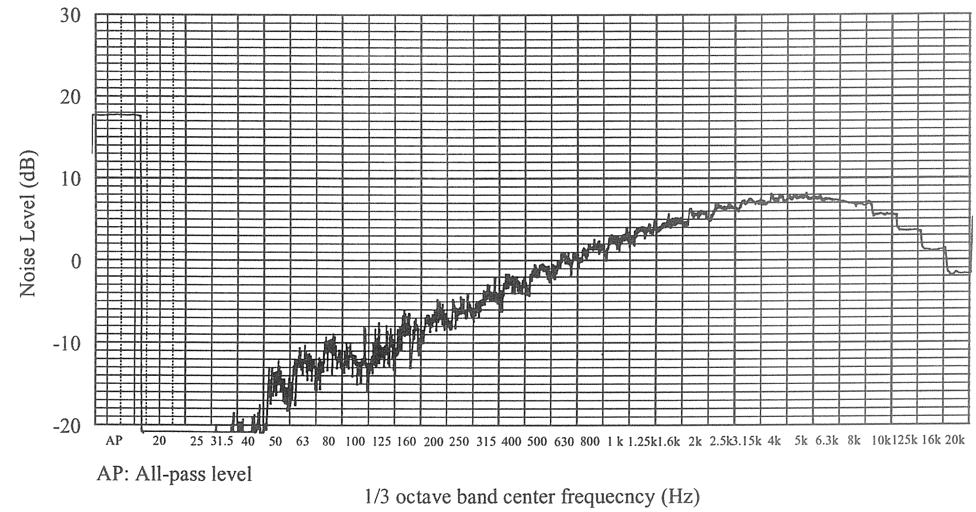
The waveform peak sound pressure level for a given measurement interval can be measured.

Noise Floor

The diagrams below show the inherent noise of the NL-18, in the frequency weighting "A" and "Flat" positions. The measurement was made with a 1/3 octave filter and a frequency analyzer.



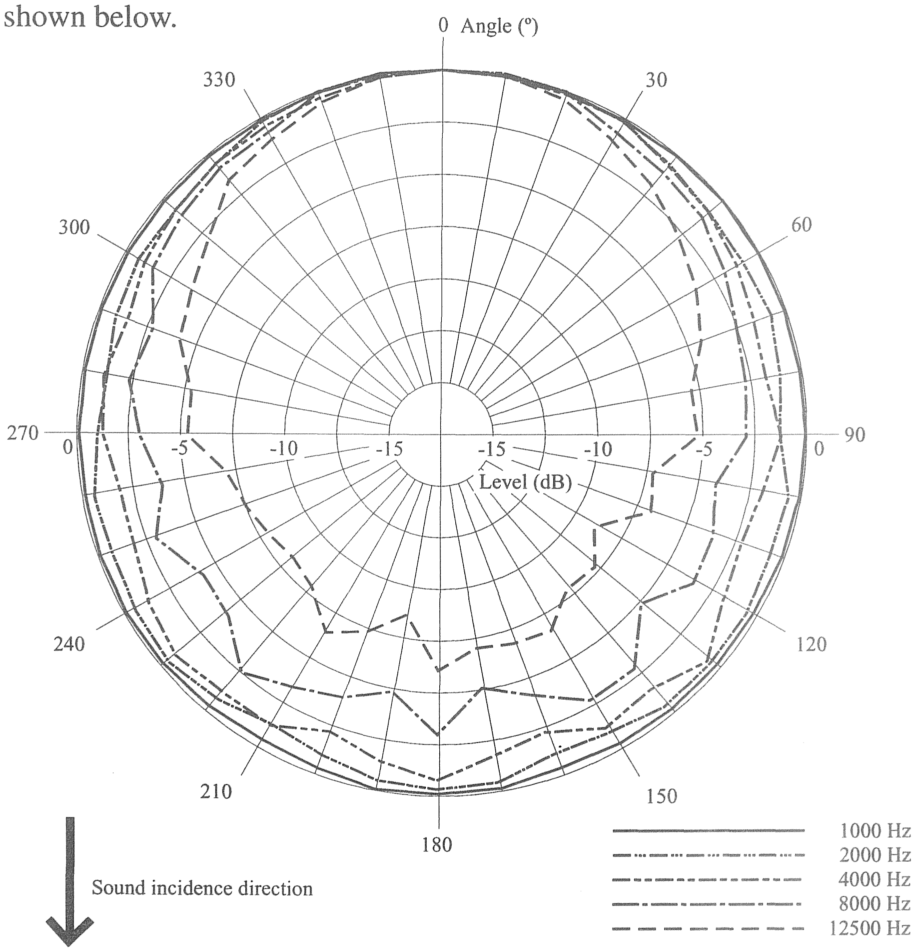
Noise floor of NL-18 in flat position

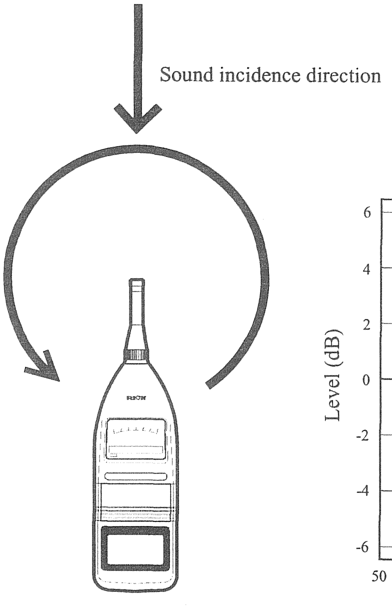


Noise floor of NL-18 in frequency weighting A position

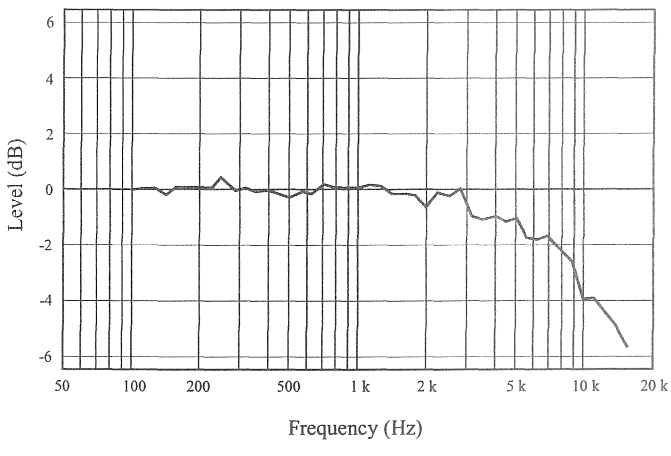
Influence of Unit Body

The directional characteristics of the NL-18 and the random incidence response are shown below.

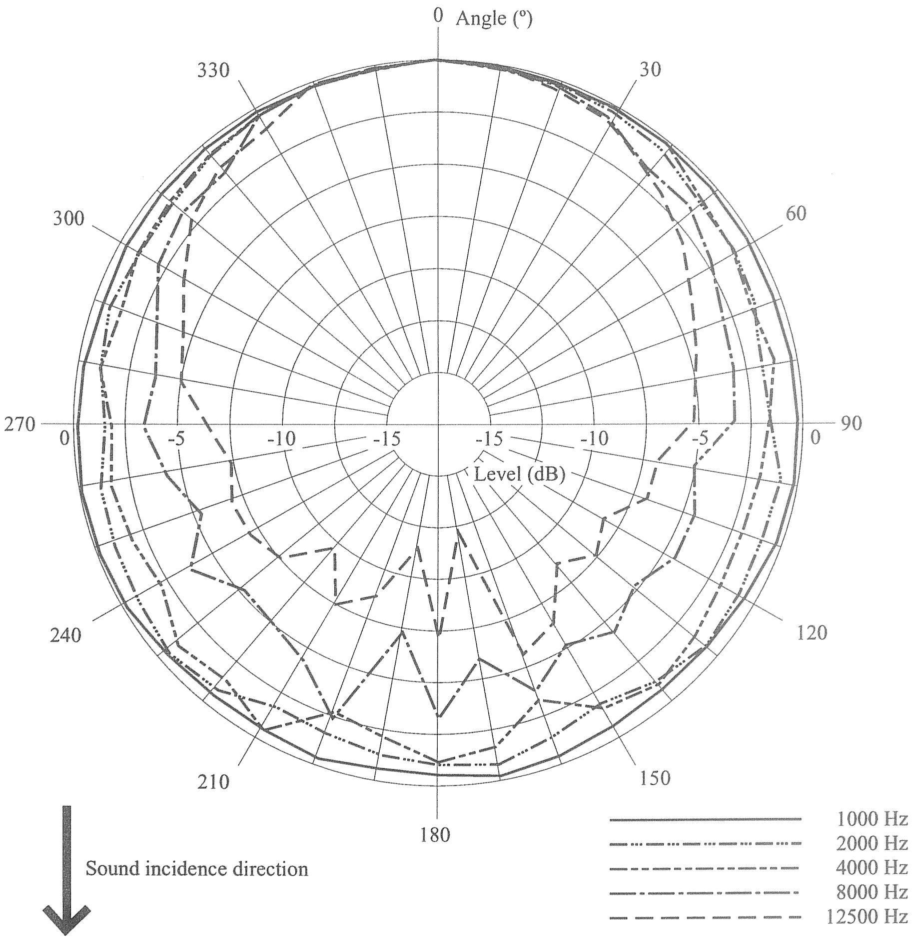




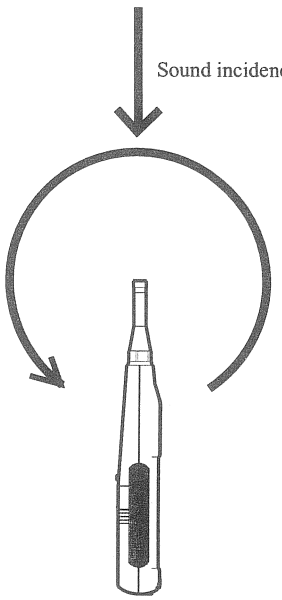
Unit rotated with microphone as center



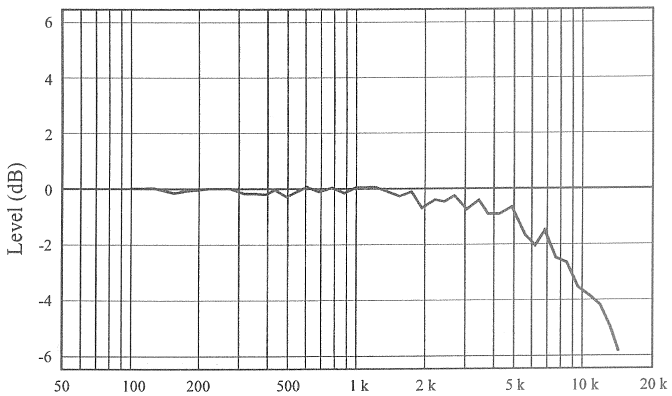
Random incidence response



Sound incidence direction



Unit rotated with microphone as center



Random incidence response

Influence of Background Noise

When measuring a certain sound in a certain location, all other sounds present at that location except the measurement target sound are background noise (also called ambient noise or dark noise). Since the sound level meter will display the combination of target sound and background noise, the amount of background noise must be taken into consideration when determining the level of the target sound.

If the difference between the meter reading in absence of the target sound and the reading with the target sound is more than 10 dB, the influence of background noise is small and may be disregarded. If the difference is less than 10 dB, the values shown in the table below may be used for compensation, to estimate the level of the target sound.

Background noise compensation

<table>
<tr><td>Display reading difference with and without target sound (dB)</td><td>4</td><td>5</td><td>6</td><td>7</td><td>8</td><td>9</td></tr>
<tr><td>Compensation value (dB)</td><td colspan="2">-2</td><td colspan="4">-1</td></tr>
</table>

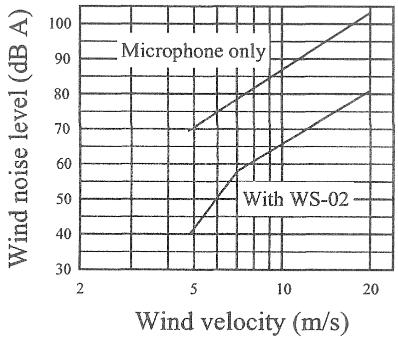
If for example the measured sound level when operating a machine is 70 dB, and the background noise level when the machine is not operating is 63 dB, the compensation value for the difference of 7 dB is -1 dB. Therefore the sound level of the machine can be taken to be 70 dB + (-1 dB) = 69 dB.

The above principle for compensating the influence of the background noise assumes that both the background noise and the target sound are approximately constant. If the background noise fluctuates, and especially if it is close in level to the target sound, compensation is difficult and will often be meaningless.

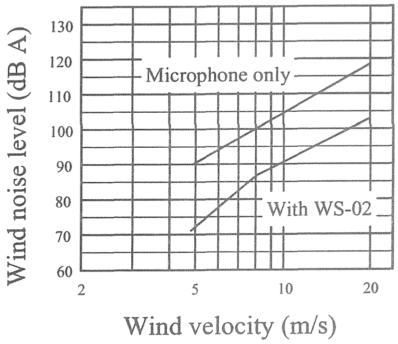
Reduction of Wind Noise by Windscreen

During outdoor measurements or measurement of ventilation devices, wind noise can falsify measurement results. To counter such problems, the supplied windscreen WS-02 should be mounted on the microphone. The characteristics of the WS-02 are shown below. The attenuation of wind noise produced by the windscreen is about 25 dB with "A" frequency weighting and 15 dB with "C" frequency weighting.

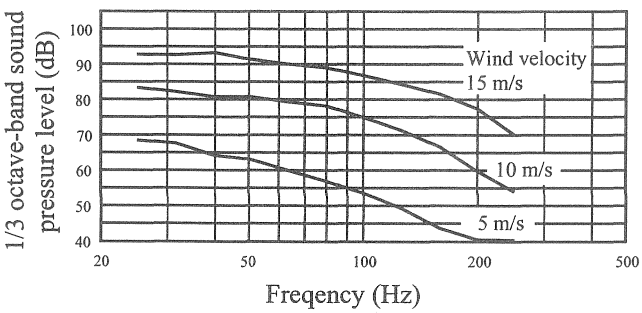
The influence of the windscreen WS-02 on the acoustic performance of the microphone is within ±1.0 dB up to 12.5 kHz, as shown in the diagram on the next page.



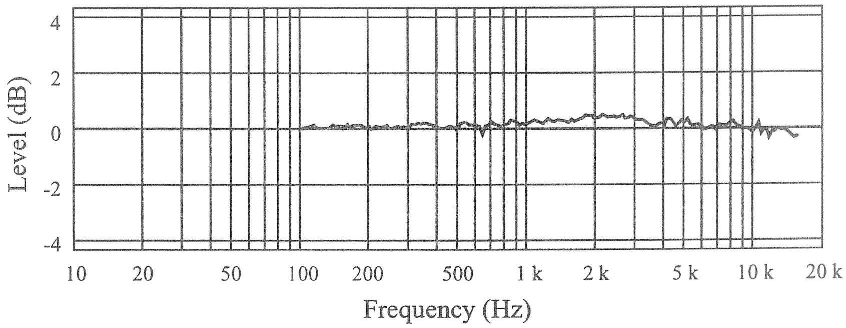
Frequency weighting A



Frequency weighting C



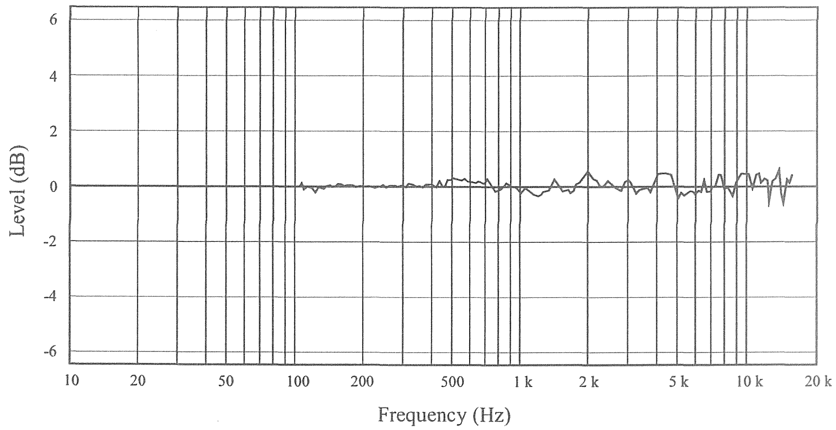
Frequency response of wind noise measured with windscreen WS-02 mounted on microphone



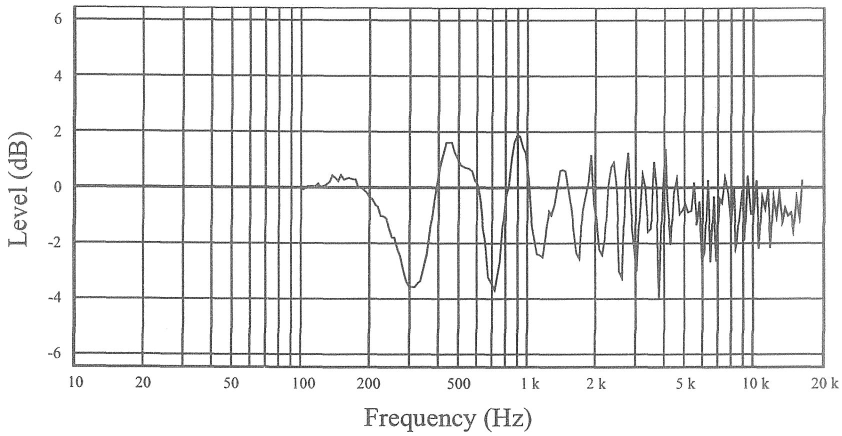
Influence of windscreen WS-02 on acoustical properties of microphone (referred to microphone response without windscreen)

Influence of Operator

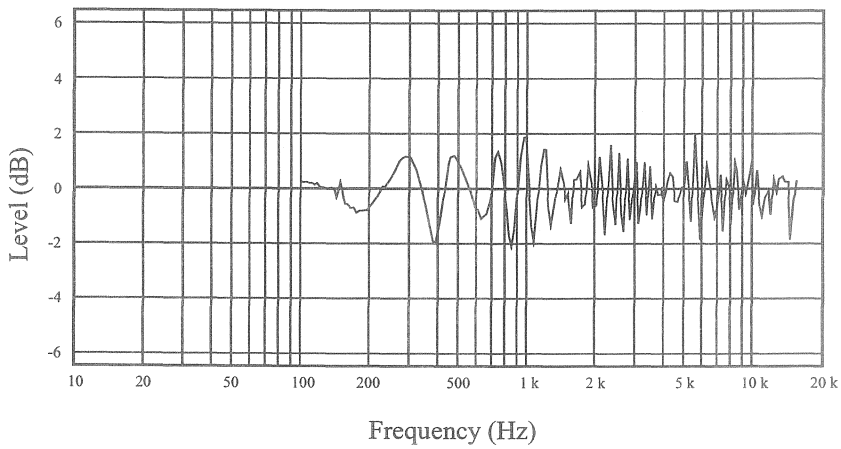
The NL-18 is designed to minimize reflections caused by the body of the unit. The charts below show the influence of the sound level meter body and the operator on the measurement.



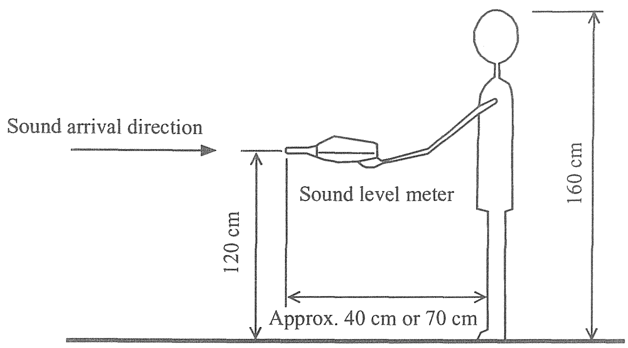
Acoustical influence of sound level meter body



Acoustical influence of sound level meter body (the distance from the top of the microphone to the operator is approx. 40 cm)



Acoustical influence of sound level meter body (the distance from the top of the microphone to the operator is approx. 70 cm)



Measurement conditions for acoustical influence of operator

I/O Connector

The I/O connector on the NL-18 serves for input of control signals and input/ output of data. It has the following functions.

- Measurement data output to printer CP-10, CP-11
  (with printer cable CC-90)
- Measurement parameter output to level recorder LR-06
  (with NL information transmission cable CC-31)
- Filter control input from level recorder LR-06/LR-04
  (with sync cable CC-91)
- Communication with a computer (RS-232C interface)
  (with interface cable CC-87E)

Filter Characteristics

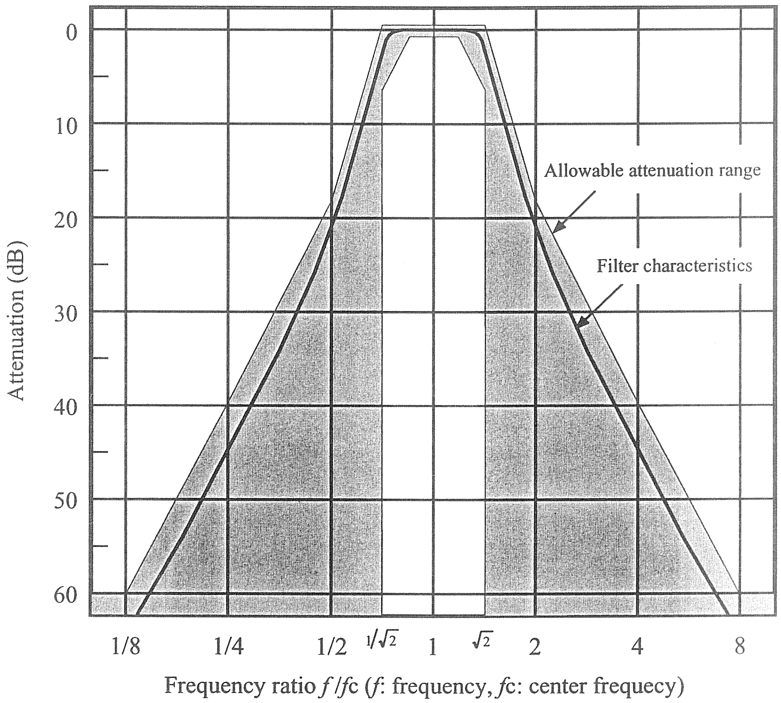
The characteristics of the filter unit NX-04 and NX-05 are shown below.

1/1 octave filter

The filter has the following 11 center frequencies (Hz):

16 31.5 63 125 250 500 1000 2000 4000 8000 16000

Attenuation characteristics correspond to JIS C 1513 - 1983 Type II



Attenuation tolerance according to JIS C 1513-1983 Type II
and 1/1-octave band filter characteristics of NL-18

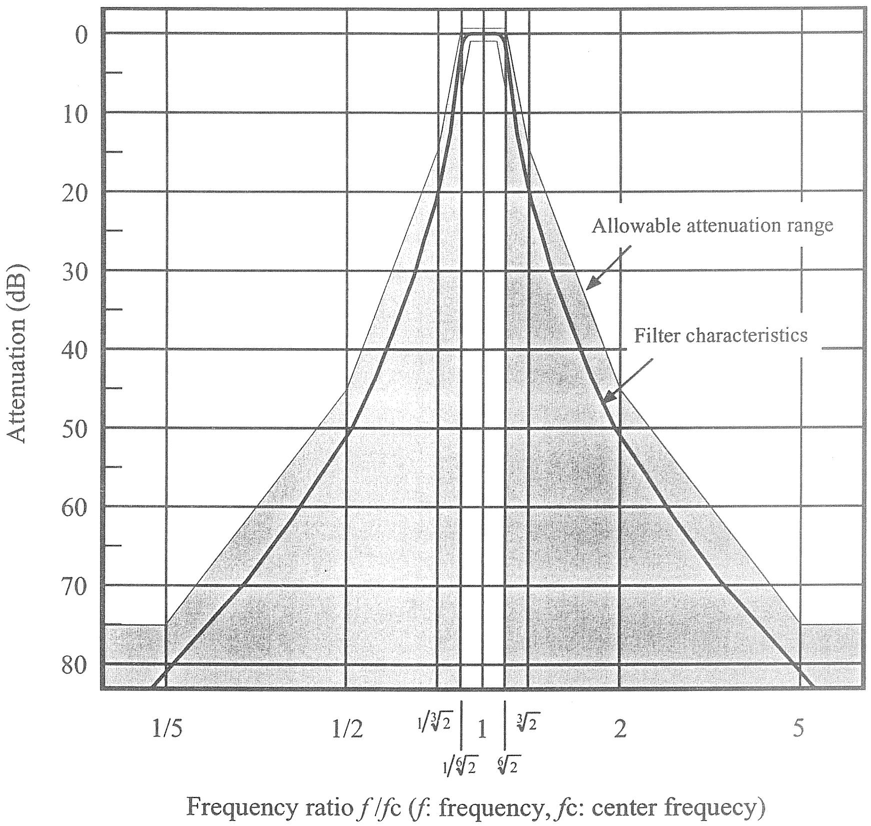
Important
The frequency range of the noise level meter NL-18 is 20 to 12500 Hz. Measurements at center frequencies outside of this range may not be reliable.

1/3 octave filter

The filter has the following 33 center frequencies (Hz):

12.5	25	50	100	200	400	800	1600	3150	6300	12500
16	31.5	63	125	250	500	1000	2000	4000	8000	16000
20	40	80	160	315	630	1250	2500	5000	10000	20000

Attenuation characteristics correspond to JIS C 1513 - 1983 Type III



Attenuation tolerance according to JIS C 1513-1983 Type III
and 1/3-octave band filter characteristics of NL-18

Important
The frequency range of the noise level meter NL-18 is 20 to 12500 Hz. Measurements at center frequencies outside of this range may not be reliable.