# Loudspeaker measurements with Audio Analyzers UPD or UPL

Application Note 1GA16_1L

Products:

**Audio Analyzer UPD**
**Audio Analyzer UPL**

**ROHDE&SCHWARZ**

# Content

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# 1 Introduction

Audio Analyzers UPD and UPL with their large variety of functions provide practically all measurement procedures required in audio technology. Thanks to the Universal Sequence Controllers UPD-K1 and UPL-B10, which are available as options, the user is able to considerably expand the range of functions to suit his particular requirements. It is possible, for instance, to add complete measurement functions - eg measurement of ohmic resistance - and to read out, convert or reload sweep results or to add new scale labelling (eg group delay). Another example is the analysis of sweep curves and the display of results in a window next to the curve. As far as softkey labelling and functions are concerned, operation of the sequence control programs is analogous to the softkey control of the UPD/UPL graphics display.

This application contains a BASIC program permitting automatic measurements of loudspeaker parameters with the aid of the features described above. The program can be run on UPD with UPD-K1 and on UPL with UPL-B10. The application software supports the following measurements or measurement sequences:

- Measurement of impedance characteristic with calculation of tuning frequency, resonance frequency and resonance impedance, DC resistance, minimum and maximum impedance, Q factor and equivalent volume.
- Measurement of sound characteristic with calculation of transmission range as well as measurement of sensitivity and peak sound pressure.
- Measurement of intermodulation products at fixed frequencies or of harmonic distortion as a function of frequency using a variety of measurement procedures and displays
- Measurement of phase linearity and group delay as well as
- Various methods for checking the polarity

The software not only yields highly accurate results for the development of loudspeakers but also supports measurements in production testing:

- Impedance- and frequency-response curves can be referenced to a "golden unit" so that a GO/NOGO decision can be made for each tested device.
- Control of measurement sequences can be stored (learned) by the program so that whole sequences can be performed by a keystroke without any further intervention being required. Thus several measurements can be combined to a series of measurements (macro operation).

The measurements and calculations described in this Application Note are in line with DIN IEC 268, part 5.

# 2 Preparations

## 2.1 Hardware and Software Requirements

The accompanying floppy disk holds BASIC program modules and one complete setup (SPEAKER.SCO), so that the application software can be run immediately under the UPD/UPL universal sequence controller (UPD-K1 or UPL-B10). As regards the hardware, an UPD/UPL of basic configuration will be sufficient. An external keyboard is required for operating the automatic sequence controller.
The following software requirements must be met:

- UPD software version 3.03 or higher, UPL software version 1.01 or higher,
- Universal Sequence Controller UPD-K1 or UPL-B10,
- configuration 3 (64-Kbyte program and 32-Kbyte data memory), for very long sweeps (more than 400 sampling points) or combined near/far-field measurements possibly configuration 5 (64-Kbyte data memory).

**Note**: *UPD and UPL are configured with the aid of service programs UPDSET and UPLSET, respectively.*

## 2.2  Software Installation

Same as the UPD/UPL software, the application software is installed with the aid of the installation program APPINST.BAT:

- quit UPD/UPL (press SYSTEM key or Ctrl F9,
- insert the supplied floppy disk,
- select floppy disk drive  (A:),
- call up the installation program (APPINST).

The installation program creates a subdirectory with the name SPEAKER (if not yet available) and then copies the BASIC program modules and setups required for the application into this subdirectory.

After the installation, the SPEAKER.SCO setup can be adapted to specific user requirements and to the external hardware (printer, monitor). See also section 2.4.2, Preliminary Configuration of Setup File.

## 2.3  Starting the Application Software

The application software is loaded and started under the universal sequence controller. So the device software has to be started first (either by switching the instrument on or by calling up the UPD/UPL at DOS level) and then the universal sequence controller is selected with key F3 (external keyboard).

***Note:***  *Before selecting the universal sequence controller make sure that the logging mode is switched off ("logging off" displayed at the very right of the command line) as otherwise commands entered in the manual mode are added to the application program and this might cause the BASIC program memory to overflow. The logging mode is switched on and off by means of key F2.*

The application software must be started from the path

>    ⇒    \SPEAKER

as it searches for all program modules in the current path. Therefore, either the path has to be changed or the application setup \SPEAKER\SPEAKER.SCO be loaded.

The path can be changed in one of the following ways:

- in the device software under "Working Dir" in the FILE panel
- in the automatic sequence controller with the BASIC command line

>    ⇒    UPD OUT "MMEM:CDIR '\SPEAKER'".

>    ***Note****: UPL OUT and UPD OUT are synonymous, the BASIC program always uses UPD OUT.*

- at the DOS level with CHDIR (via the SHELL of the automatic sequence controller)

The BASIC program SPK.BAS can now be loaded and started. To do so the following entries have to be made:

- press the LOAD softkey,
- enter SPK  (confirm with "return"),
- press the RUN softkey.

## 2.4  Configuring the Application

### 2.4.1  Setup and Configuration Files

The application software uses two different sources for device setting:

SPEAKER.CFG: Configuration file holding all user entries (see section 3.4, Entry of Parameters) or default values on initial call-up. When the application is started, this file is loaded; when the application is left using the EXIT softkey (see section 7, Terminating the Application) the file is stored with updated values so that the entries made by the user during the last run are reinstated on starting the application again. If as an exception user entries should not be stored, the application should be quit with CTRL BREAK.

SPEAKER.SCO: Setup file used as the basis for all commands of the application software. Most of the settings in this file have to remain unchanged or are overwritten by the application software. A few settings only may be (or have to be) updated by the user. The program reads the setup but does not change it. The final version of the setup may be protected against inadvertent overwriting by assigning the attribute READ ONLY.

*Note:* *Alternative configuration files can be stored and loaded under the menu item UTILITY -> CONFIG. This capability is particularly important when macros are used (see section 6, Combining Several Measurements).*

The SPEAKER.SCO setup also includes settings that are identical for all procedures and measurements:



FIG 1: Generator, analyzer and display panel of SPEAKER.SCO setup

### 2.4.2 Preliminary Configuration of Setup File

The following modifications of the SPEAKER.SCO setup file are permissible:

- selection of printer interface (SCREEN HARD COPY in OPTIONS panel),
- defining a maximum generator voltage for protecting DUTs,
- driving an external monitor (Extrn Disp in OPTIONS panel),
- changing the X axis from log to linear (Spacing LIN POINTS in GENERATOR panel and X-AXIS->Spacing LIN in DISPLAY panel),
- changing the Y1 axis from lin to log (TRACE A->Spacing LOG in DISPLAY panel), applies to impedance measurements only,
- selection of commands to be displayed in the status panel ("tick off"),
- comments in plain text in the graphics display (DISPLAY panel) and info text (FILE panel)
- comments in plain text for printout (dialog box of hardkey HCOPY),
- defining tolerance curves/values and checks (LIMIT CHECK in DISPLAY panel).

The application software is tailored to the audio range, ie to the 25-kHz generator and a 22-kHz analyzer. In the UPD instruments for higher frequencies may also be selected but in this case the following aspects have to be considered:

- the lower limit frequency of the analyzer is higher,
- the resolution of the 8k FFT is only 41.67 Hz (100-kHz analyzer) or 125 Hz (300-kHz analyzer),
- the measurement speed is lower.

With the preliminary configuration terminated, the setup has to be stored (as complete setup) under the same name and may be assigned the attribute READ ONLY.

*IMPORTANT*: Only the listed settings may be modified, as otherwise proper function of the software cannot be guaranteed.

## 2.5  Converting the Setup for Software Updates

If the device software is updated, the SPEAKER.SCO - and all other setups - may require to be converted. This is done automatically when the setup is loaded but to avoid unnecessary delays during loading, SPEAKER.SCO should be stored as converted setup. This can be done in two ways:

- at the DOS level by calling up the batch converter:

  ⇒ UPD_CONV \SPEAKER\SPEAKER.SCO        or
  ⇒ UPL_CONV \SPEAKER\SPEAKER.SCO,

- at the UPD/UPL level by loading and storing SPEAKER.SCO.

IMPORTANT: with setups set to READ ONLY, the "r" attribute has to be deleted at the DOS level:

  ⇒ ATTRIB  -r \SPEAKER\SPEAKER.SCO

# 3  Operating Concept

## 3.1  Overview

The application is operated by means of softkeys which are organized in 4 hierarchical levels. When the application software is started, the first (highest) softkey level is displayed; pressing a softkey triggers an action and/or causes another softkey level to be entered. Each level may have up to eight active softkeys; unassigned softkeys are blank. Same as in the UPD/UPL device software, function keys F5 to F12 correspond to softkeys #1 to #8.

Each time a softkey is pressed, the displayed softkey are cleared. No entries can be made while the application program carries out setting sequences or calculations or waits for the end of a measurement. The softkeys are blank. Only after the softkeys have been assigned new (application) functions can further entries be made.

## 3.2  Softkey Levels

Functions of the four hierarchical softkey levels:

- Level 1 (main level): selection of measurement; implicit loading of a specific measurement module
- Level 2: selection of measurement mode (frequency sweep or FFT noise; see section 4, Measurement Modes) with implicit (re)-loading of SPEAKER.SCO setup;
- Level 3: entry of parameters and start of measurement;
- Level 4: post-processing and restart of measurement.
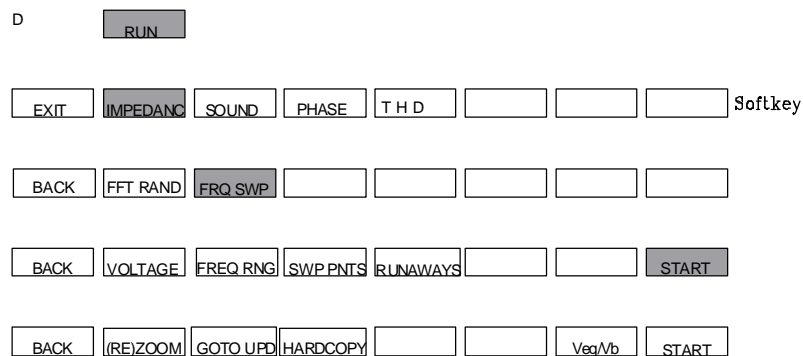
FIG 2: Operation of application software via softkey menus

## 3.3  Softkeys Common to all Levels

Two of the altogether eight softkeys are assigned the same function at all levels and are always located at the same position:

- the lefthand softkey (#1, F5) is used for returning to the next higher level (BACK); at the top level  the application is quit (EXIT; see section 7, Terminating the Application),
- the righthand softkey (#8, F12) is only active at levels 3 and 4 and used for starting the measurement; at the end of the measurement the application is always at level 4.

## 3.4 Entry of Parameters

All parameters are entered at level 3 and initiated by softkeys. Strings, integer and floating-point numbers are entered in a dialog, a 1-out-of-N selection is made at another softkey level.

For dialog entries the following applies:

- The designation of the parameter (eg voltage) is displayed in the entry line, possibly together with the unit (eg V).
- The currently set value is displayed at the right of the cursor. It may be overwritten or confirmed by means of the RETURN key.
- Floating-point numbers are entered in the specified unit; only the value is entered.
- If the permissible upper or lower limit is exceeded, the entry is limited to this value or a new entry is requested.

For 1-out-of-N entries the following applies:

- Selecting a 1-out-of-N entry opens up a 1-out-of-N selection level at which selectable values are assigned to the softkeys.
- When the desired softkey is pressed the selected value becomes valid immediately and the 1-out-of-N selection level is closed.

Since the parameters to be entered in this way are the same for most of the measurements, they will be explained in brief below. Special parameters shall be dealt with together with the respective measurement. The mentioned measurement modes FFT noise and frequency sweep are explained in detail in section 4.

- VOLTAGE (F6, level 3): total voltage (FFT noise) or sinewave voltage (frequency sweep), at the same time DC voltage for measuring the DC resistance.

- FREQ RNG (F7, level 3): Lower Freq (in Hz) and Upper Freq (in Hz): frequency range within which sinusoidal lines are generated (FFT noise) or sweep points set (frequency sweep). With FFT noise mode, reducing the frequency range also reduces the number of sinusoidal lines and therefore shortens the time required for generating the noise signal. If frequency sweep is used the resolution is improved (constant number of sweep points).

- SWP PNTS (F8, level 3): number of sweep points (frequency sweep only). Reducing the sweep points shortens the sweep time, an increase improves the resolution.

- FFT SIZE: number of FFT lines determining the FFT resolution and therefore the FFT noise. Reducing the lines causes the resolution to deteriorate but shortens the time for generating the noise signal and the FFT measurement time.

## 3.5 Common Softkeys for Post-Processing

The softkeys for post-processing are at softkey level 4. Some of them are available for all measurements and always assigned the same function (and label):

- Softkey #2 (F6): o <--> O changes from partial to full graphics display and vice versa.
- Softkey #3 (F7): GOTO UPD changes temporarily to the UPD operating software (see section 7.1, Change of Task Application/User Interface)
- Softkey #4 (F8): HARDCOPY outputs the screen content to a printer or file (see section 7.2.2, Printout during Application Run)

## 4  Measurement Modes

### 4.1  Modes of Frequency Response Analysis

Measuring the frequency response of various parameters is particularly important in development and production testing. Such parameters are for instance:

- electrical transfer function of crossover units,
- acoustic transfer function of loudspeakers,
- impedance characteristic,
- sound pressure characteristic,
- phase characteristic and group delay,
- distortion.

These parameters can be measured by the frequency sweep of a sinewave signal and displayed in a tabulated or graphical form. Each curve may contain of up to 1024 single measurements with linear or logarithmic scaling as required.

Frequency sweeps are very time-consuming because of the generator setting time, DUT settling time, measurement time and, for acoustic measurements, the signal travel time involved. Particularly at low frequencies, settling time and measurement time may be too long when it comes to alignments or production testing.

All measurements based on voltage measurements may also be carried out with a noise generator and an FFT analyzer. The (quasi)-noise generator of the UPD/UPL uses discrete sinusoidal lines and may be synchronized to the FFT, ie the sinusoidal audio signals produced by the generator are of the same frequency spacing as the FFT (called FFT noise in the text below). Consequently the measurement signal at the display edges of the FFT can be continued without any transitions that no window function is required for suppressing the FFT continuation error and an optimum frequency and level accuracy of the FFT is obtained.

Measurement time and frequency resolution depend on the selected FFT size or on the frequency stepwidth of the noise generator.
The smallest step of the 25-kHz generator is 5.86 Hz. This is therefore the maximum permissible resolution for the FFT analyzer although a much better resolution could be obtained in the analyzer (with the aid of ZOOM FFT). It is advisable to use 8 k FFT where the measurement speed still allows several measurements per second to be made. The measurement speed can be further increased by reducing either the frequency resolution or the FFT size.

At the generator side, generation and phase optimization of the noise signal involves much more computation work than the generation of a simple sinewave signal. If all available sinusoidal lines are used, the noise signal is available after about 1 minute. However, this time is only required once and any number of highly accurate measurements can be carried out thereafter. By reducing the FFT size (analyzer) or the frequency range (generator) the number of discrete sine tones can be reduced and the signal generation time drastically shortened.

***Note***:  *The highest resolution (2.9296875 Hz at 8 k zoom FFT) makes only sense in the UPL, as the UPD is not able to track a resolution of < 5.859375 Hz because of its higher internal generator clock rate.*

The frequency sweep and the FFT noise mode each have their pros and cons so that both modes are employed (as far as possible) for the measurements described in the Application Note.

Another measurement mode implemented particularly for transmission range measurements is the swept burst measurement. The measurement time is longer than in the normal (sinusoidal) sweep mode, but above a room-dependent lower frequency limit, echofree measurements can be performed outside anechoic rooms.

## 4.2  Frequency Sweep

### 4.2.1  Advantages

- Logarithmic frequency scaling is possible so that a very good frequency resolution is obtained at low frequencies. FFT uses linear scaling only.
- The frequency resolution can be improved without any restrictions by reducing the frequency range; with FFT the smallest step is fixed to 5.86 Hz (UPD) or 2.93 Hz (UPL).
- The generator setting time for each sweep point is very short while a time-consuming noise-signal generation is required with FFT noise.
- Frequency sweep is possible with all measurements,  FFT noise can be used for voltage measurements only.

### 4.2.2  Setup and Program Settings

When selecting the frequency sweep mode (see section 4.6,  Selecting the Measurement Mode) the SPEAKER.SCO setup with the following settings becomes active:

```
                        RMS                          GEN RUNNING
                                                     ANL 1:SNGL 2: OFF
        CH1 | 298.47 V/Vr |   | Off |   | Off |      SWP SNGL RUNNING
        CH2 |     Off      |   | Off |   | Off |      Jul 01 1994
                                                     Fri 10:34:03

  ┌──── GENERATOR ────┐    ┌──── ANALYZER ────┐    ┌──── DISPLAY ────┐
  · Ref Volt   1.0000 V     ▲                    ▲   ■ OPERATION  CURVE PLOT
                            √ FUNCTION − RMS & S/N      · Mode       DEL BEF WR
  √ FUNCTION − SINE         · S/N Sequ    OFF          · User Label OFF
  · Frq Offset OFF          · Meas Time   AUTO
  · DC Offset  OFF          · Unit Ch1    V/Vr         ■ TRACE A — FUNC CH1
  · Low Dist   OFF          √ Reference   VALUE:        · Unit       V/Vr
  · SWEEP CTRL AUTO SWEEP                 20.000 mV     √ Reference  VALUE:
  · Next Step  ANLR SYNC    · Notch(Gain)OFF                        20.000 mV
  · X Axis     FREQ         · Filter     OFF          · Scale      AUTO ONCE
  · Z Axis     OFF          · Filter     OFF          · Spacing    LIN
  FREQUENCY                 · Filter     OFF
  √ Spacing    LOG POINTS   · Filter     OFF          ■ TRACE B — OFF
  √ Start      10.000 Hz    · Fnct Settl OFF
  √ Stop       20000  Hz                              ■ X AXIS —— FREQ
  √ Points     400_                                   · Unit       Hz
  · Equalizer  OFF          ■ POST FFT    OFF         · Reference  1000.0 Hz
  √ VOLTAGE    6.0000 V     ■ SPEAKER — OFF           · Scale      AUTO      ▼
```
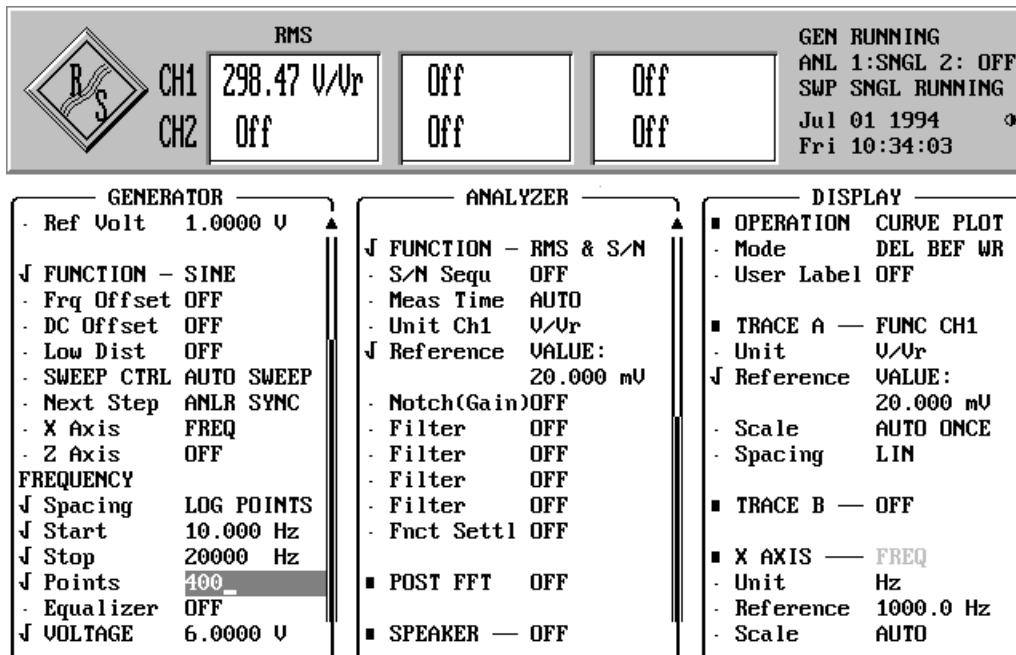
FIG 3: Example of SPEAKER.SCO setup for frequency sweep mode

Values for start, stop, points and voltage are generated from user entries or taken from the SPEAKER.CFG configuration file.

## 4.3  FFT Noise

### 4.3.1  Advantages, Limits and Main Fields of Application

- The pure measurement time - without signal generation time - is very short (online response).
- An increase (doubling) of the frequency resolution increases the measurement time only insignificantly.
- Up to 3712 measurement points  - 7424 in the UPL - are possible (1024 in the sweep mode) so that a higher frequency resolution is possible over the entire sound frequency range with linear scaling.

In the present application the FFT noise mode can be used for measuring the impedance and sound pressure characteristic. Main fields of application are:

- production testing (go/nogo tests),
- adjustments,
- sound level measurements in not completely anechoic rooms.

### 4.3.2  Setup and Program Settings

Selecting the FFT noise mode (see section 4.6, Selection of Measurement Mode) activates the SPEAKER.SCO setup with the following settings:

```
        RMS via FFT                              GEN RUNNING
   ___                                           ANL 1:CONT 2: OFF
  /R S\  CH1 2.0164 V/Vr    Off        Off       SWP OFF
  \___/                                          Jul 01 1994      ☾
         CH2 Off            Off        Off       Fri 10:41:18

 ┌─── GENERATOR ───┐   ┌─── ANALYZER ───┐   ┌─── DISPLAY ───┐
 - Channel(s) 1      ▲  √ FUNCTION — FFT    ▲  ■ OPERATION  SPECTRUM
 √ Output    BAL  XLR    - Unit Ch1    V/Vr    - Mode       DEL BEF WR
 √ Impedance  600 Ω      - Reference   VALUE:   - User Label OFF
 - Common     FLOAT                   1.0000 V
 - Max Volt   24.000 V   - Notch(Gain)OFF       ■ TRACE A — FUNC CH1
 - Ref Freq   1000.0 Hz  √ FFT Size    8192     - Unit       V/Vr
 - Ref Volt   1.0000 V   √ Window   RECTANGULAR √ Reference  VALUE:
                         - Avg Mode  EXPONENTIAL              20.000 mV
 √ FUNCTION — RANDOM     - Avg Count   1        - Scale      AUTO ONCE
 - DC Offset  OFF        - Zooming     OFF      - Spacing    LIN
 - Domain     FREQ       - Start    0.0000 Hz
 √ Spacing   ANLR TRACK  √ Stop     21750   Hz  ■ TRACE B — OFF
 √ Equalizatn WHITE      √ Resolution 5.8594 Hz
 √ Lower Freq 11.719 Hz  - Meas Time  0.1707 s  ■ X AXIS — FREQ
 √ Upper Freq 20004  Hz                         - Unit       Hz
 √ VOLT PEAK  6.0000 V   ■ SPEAKER — OFF        - Reference  1000.0 Hz
 √ VOLT RMS   2.0271 V                          - Scale      AUTO      ▼
 └─────────────────┘   └────────────────┘   └───────────────┘
```
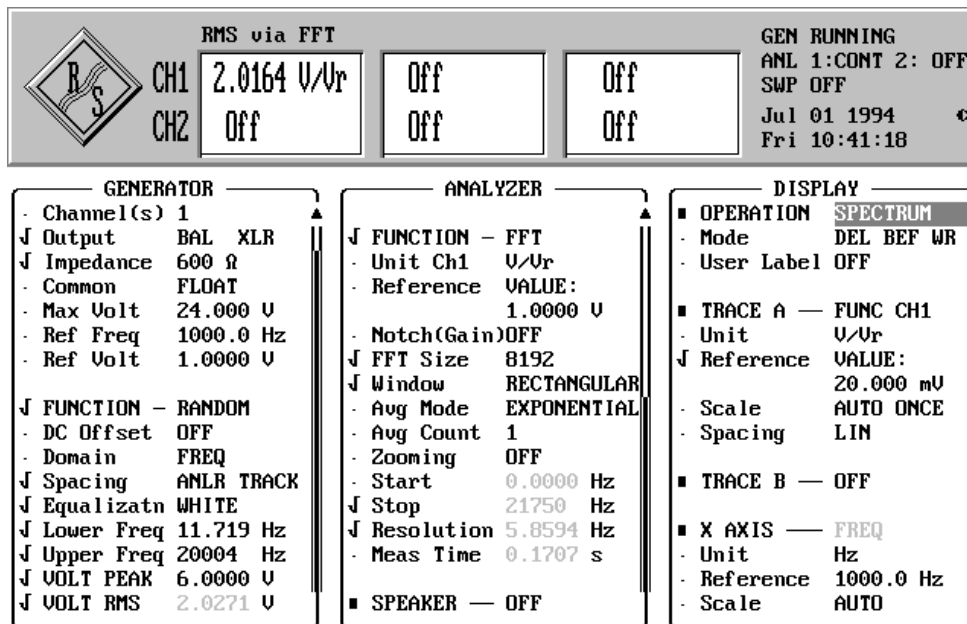
FIG 4: Example of SPEAKER.SCO setup for FFT noise mode

The values for Lower Freq, Upper Freq, VOLT RMS and FFT size are generated from user entries or taken from the configuration file SPEAKER.CFG.

Remarks:
- If an entry is made in the GENERATOR panel which requires a regeneration of the noise signal, the generator starts the very time-consuming noise signal computation (GEN BUSY in the status display). Noise-signal generation can be aborted by a keystroke in the case of manual control but this is not possible when the universal sequence controller is used.
- "Window RECTANGULAR" switches off the window function.
- "Zooming ON" (with zoom factor 2) is only possible in the UPL, as the UPD generator is not able to set frequency resolutions below 5.8584 Hz.

## 4.4 Swept Burst Measurement

In this mode, not a continuous but a burst sinewave signal is swept, ie varied in frequency. A whole number of periods is measured, the burst duration is one period longer than the measurement time. This mode is used for measuring the transmission range as with this measurement reflections in the room can be eliminated by simple means.
Between the echo signal delay and the signal delay only the direct measurement signal is present at the microphone. This period defines the lower frequency limit for an echofree operation: all frequencies with a shorter signal period can be measured without any echoes. The procedure is described in detail in section 5.2.6, Frequency Response of Sound-Pressure Level and Transmission range.

### 4.4.1  Advantages and Main Fields of Application

- Echofree measurements above a certain lower frequency limit also outside anechoic chambers
- the measurement is not impaired by the echo of the preceding sweep point
- short burst pulses are easier to support than continuous sinewave signals.

Because of the longer measurement time compared with those of other modes and the susceptibility to interference, this mode can only be used in development laboratories.

### 4.4.2  Program Settings

Since the burst sweep cannot be directly selected as a mode in the device software, each sweep point has to be set by the application software:

- Select SINE BURST as a generator signal.
- Set analyzer to a special RMS mode (TRIGGERED) where the measurement is automatically triggered on detection of the signal level.
- Same as with the sinewave sweep, vary the frequency depending on the values set for start, stop, points and spacing.
- The measurement time is a multiple of one period, but at least one period. Towards the upper end the number of periods is limited such that the measurement is terminated before the first echo arrives.
- The burst duration is one period longer than the measurement time and also an integer (to avoid DC components).
- The burst-on delay is constant and should compensate the internal measurement delay.
- The input level range is fixed and maintained for all sweep points. An automatic routine defines the level range so that the reference frequency (SNS FREQ) is optimally driven.
- The trigger threshold at which the RMS measurement is started can be selected (TRGLEVEL softkey). The threshold should be selected so that it is exceeded by all sweep points but sufficiently away from ambient noise. Values around 1% are recommended.

## 4.5 GO/NOGO Tests

For applications in production, the impedance characteristic and transmission range of a reference loudspeaker can be recorded, tolerance values defined and DUTs checked for compliance with set tolerances (GO/NOGO tests).

The measured characteristic of this "golden unit" is stored as a trace and loaded into the display panel as a reference trace for the DUTs. The tolerance curves are then displayed as horizontal lines at a specified distance above and below the 0-dB line. The advantage of this method is that deviations from the ideal trace (0-dB line) can be directly read in dB. An ideal DUT would produce a horizontal line of exactly 0 dB.

If the DUT is within the specified tolerances (limits), "DUT passed" is displayed. If at least one measured value is out of tolerance, "DUT failed" is displayed. If a macro is used, the macro is stopped and the instrument waits for a keystroke.

At the end of the measurement, the reference curve is stored by pressing the STO REF softkey and entering a file name. The reference curve can be called up and tolerance limits entered with the REF SPK softkey.

## 4.6 Selecting the Measurement Mode

The measurement mode is selected at softkey level 2.

Impedance measurement:
- Softkey #2 (F6, FFT RAND) selects FFT noise and
- Softkey #3 (F7, FRQ SWP) the frequency sweep for determining the impedance characteristic.

Sound measurement:
- Softkey #2 (F6, FFT RAND) selects FFT noise and
- Softkey #3 (F7, FREQ SWP) the frequency sweep for determining the transfer function.
- Softkey #4 (F8, SNS RAND) selects FFT noise and
- Softkey #5 (F9, SNS SNGL) a fixed frequency for determining the sensitivity.
- Softkey #6 (F10, BRST SWP) selects the burst sweep for determining the transfer function.

Phase measurement:
- Softkey #2 (F6, SINGLE) selects a fixed frequency and
- Softkey #3 (F7, FRQ SWP) the frequency sweep for determining the phase.

THD(N) measurement: ZOOM FFT
- Softkey #2 (F6, THD TOTL) selects the THD measurement for *all* harmonics (d2 to d9) for determining the distortion; display as a BARGRAPH (for single measurements) or sweep trace.
- Softkey #3 (F7, THD d2) selects the THD measurement for the *first* harmonic for determining the distortion; display as a BARGRAPH (for single measurements) or sweep trace.
- Softkey #4 (F8, THD d3) selects the THD measurement of the *second* harmonic for determining the distortion; display as a BARGRAPH (for single measurements) or sweep trace.
- Softkey #5 (F9, THD+N) selects the THD+N measurement (harmonics+noise) for determining the distortion; display as a BARGRAPH (for single measurement) or sweep trace.
- Softkey #6 (F10, POST FFT) selects also the THD+N measurement, but every test point is displayed numerically or as an FFM spectrum.
- Softkey #7 (F11, ZOOM FFT) sets the FFT measurement function and permits all available FFT sizes and zoom factors to be entered. Each test point is displayed as an FFT spectrum.

Upon this selection, SPEAKER.SCO is (re-)loaded and required settings are made. The data in the configuration file and user entries made in the application program since program start (see section 3.4, Entry of Parameters) are taken into account.
User entries made during post-processing (see section 7.1, Change of Task, Application/User Interface) are overwritten.

## 5  Measurements

The SPEAKER application software covers a variety of loudspeaker measurements which are displayed in the softkey line (softkey level 1).

- IMPEDANC for measuring all relevant impedance values and corresponding frequencies and for determining associated parameters.
- PHASE for measuring phase linearity and detecting polarity reversal.
- SOUND for measuring sound pressure characteristics and loudspeaker sensitivity.
- THD for measurement and display of THD+N, THD total, d2, d3, or similar

Pressing the START softkey (F12) triggers the selected measurement and the described measurement procedure.

## 5.1  Impedance and Associated Parameters

The impedance curve of a loudspeaker chassis or system gives information on important parameters which sufficiently characterize the response at low frequencies. The parameters include the quality factor $Q_t$, resonance frequency $f_r$ and equivalent volume $V_{eq}$. Equations used in this Application Note are in line with DIN IEC 268, part 5.

Since UPD and UPL comprise a DC generator and a DC measurement function, the DC impedance can be measured without any additional equipment being required.

The impedance measurement is selected with the IMPEDANC softkey (F6).

### 5.1.1  Test Setup

For measuring the magnitude of the (complex) impedance of a loudspeaker array or system, the latter is connected to the XLR output *Channel 2* of the generator (+ and -) and the voltage drop V1 is measured at the generator output (via GEN coupling).
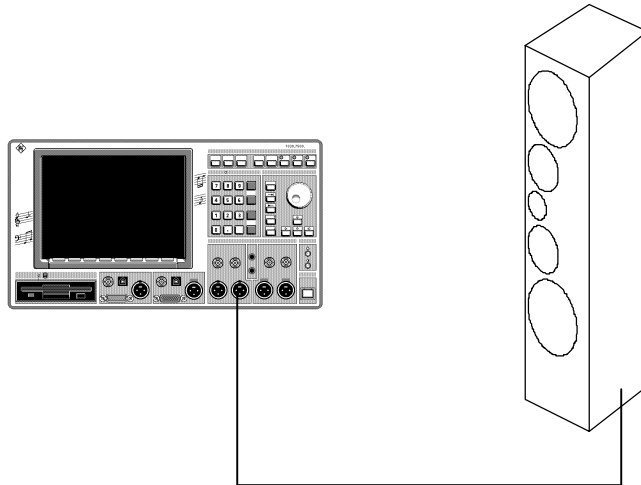


FIG 5  Test setup for impedance measurement

The open-circuit generator voltage $V_q$ being known, the following voltage ratio is obtained:

$$U_q \, / \, |Z+R| \; = U_1 \, / \, |Z|$$

Because of the complex impedance, the equation cannot be solved for Z, but the generator may be used as a current source for $|Z| \ll R$

$$I_q = U_q \, / \, R$$

so that the following applies:

$$|Z| \; = U_1 \, / \, I_q$$

For impedances in loudspeakers, the condition $|Z| \ll R$ is fulfilled if a generator with a source impedance of 600 Ω is selected. Since this source impedance is only available at the XLR outputs, the BAL outputs have to be used.

If the value for

$$I_q = U_q \, / \, 600 \; \Omega$$

is entered as reference value for the RMS or DC measurement and V/Vr is selected for the display, the measured value for Z can be directly read off in Ω.

## 5.1.2  Special Notes on Entry of Parameters

With the aid of the RUNAWAYS softkey (F9, level 3) it can be determined how many outliers (runaways) are tolerated by the algorithm for extreme value determination (resonance and tuning frequency).

In this case the relative maximum or minimum values have to be determined.  For instance, when searching for the maximum it may happen (particularly with very flat impedance curves) that a lower value is measured although the curve slightly rises. To avoid misinterpretations, the algorithm responds only if the following N RUNAWAYS show a decrease.

N is recommended to have a value of 2.  Higher values extend the search time and may even mask runaways. In the case of frequency sweeps lower values may be sufficient (even 0) but this has to be determined empirically.

Under the REF SPK softkey the GO/NOGO test mode can be switched on and off, the trace of the reference unit loaded and tolerances entered. With this softkey, OFF (no GO/NOGO test), TRACE A (GO/NOGO test only) or TRACE B (impedance curve on trace A, GO/NOGO test on trace B) can be selected.

### 5.1.3 DC Impedance

The DC impedance $R_{DC}$ is required for calculating the quality (see section 5.1.8, Q Factor). The measurement is carried out with the DC function.

The offset which can be set with almost all generator signals is used as the DC signal; the AC signal level is set to 0. The low-distortion generator has to be switched off for the offset function to become active. A DC voltage of, for instance, 6.0 V yields a current of 10 mA which is entered as reference value in the analyzer panel.
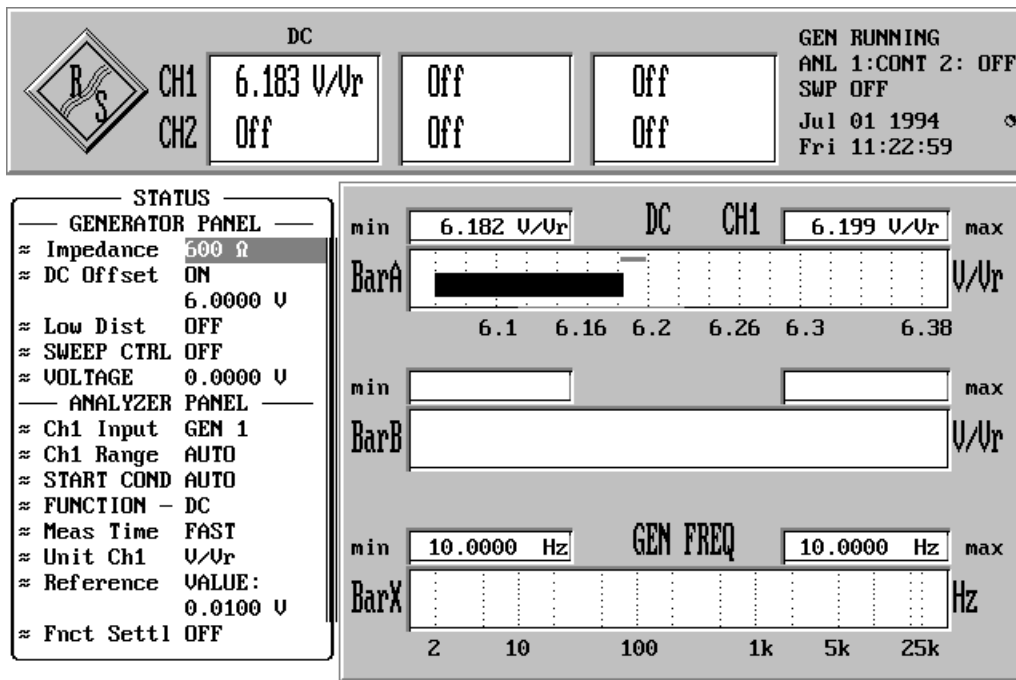


FIG 6: Settings for DC impedance measurements

### 5.1.4 Impedance Frequency Response

The impedance frequency response is required for documentation and for determining all subsequent parameters (see sections 5.1.5 to 5.1.9). According to DIN IEC 268 it is to be determined at least in the range from 20 Hz to 20 kHz. Either logarithmic frequency sweep or (linear) FFT noise may be used for the measurement. Results should in any case be displayed logarithmically. If there are no special reasons for selecting FFT noise (eg online response), frequency sweep should be given preference that in this case a smaller number of measurement points is required because of the logarithmic scaling and uncertainty of the noise signal at higher frequencies. Furthermore, the small number of sweep points reduces the program running time for subsequent search procedures.

For obtaining the maximum measurement speed, the frequency measurement is switched off. Measurements are made in 1 channel only and the (faster) function generator is selected (low-distortion generator not active). The measurement time can be further reduced by selecting AUTO FAST (however with a certain loss in accuracy).
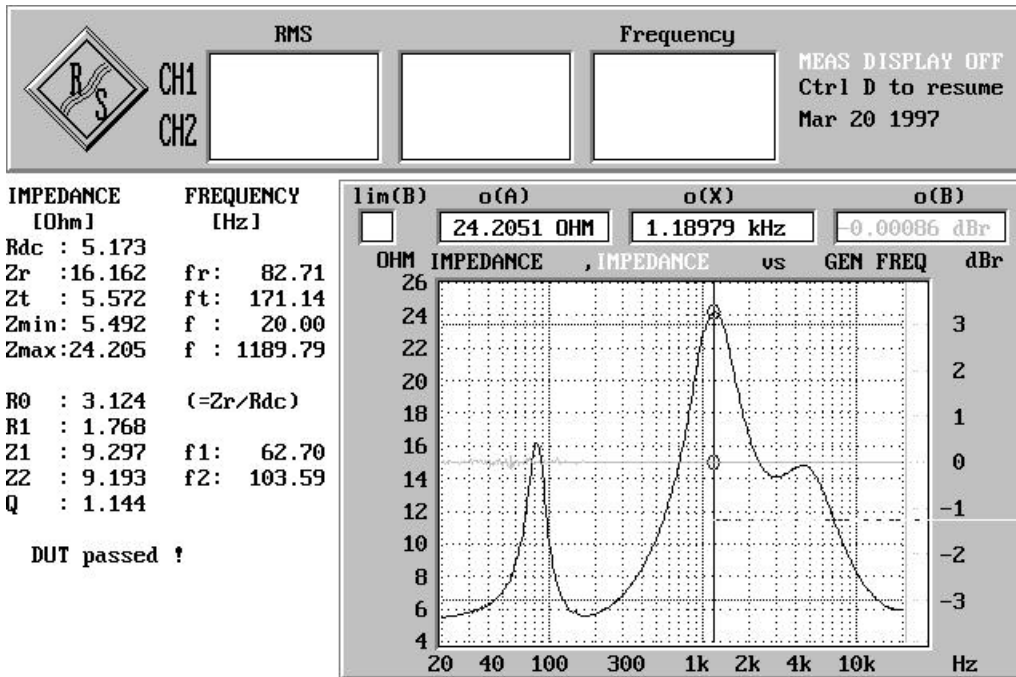
FIG 7: Impedance characteristic and evaluation of a multi-path loudspeaker
(Trace B shows the impedance characteristic referenced to a "golden unit")

### 5.1.5 Resonance Frequency and Resonance Impedance

The resonance impedance $Z_r$ is the first main maximum in the impedance curve at the resonance frequency $f_r$. The two quantities can be directly read from the impedance curve with the aid of a graphics cursor. In the program they are obtained by scrolling up through the obtained impedance value array. The maximum is considered found if N RUNAWAYS +1 (see section 5.1.2, Special Notes on Entry of Parameters) are found in successive values in the descending order.

### 5.1.6 Absolute Minimum and Maximum Impedance

The absolute maximum of impedance $Z_{max}$ is normally not identical with the resonance impedance. Using softkeys

     ⇒     Cursor --> set to --> max A

the (active) graphics cursor is set to the maximum and the impedance and associated frequency can be read from the display. Correspondingly, the absolute minimum is determined with softkeys

     ⇒     Cursor-->set to-->min A

This method is also used in the program.

### 5.1.7 Tuning Frequency

The tuning frequency is the frequency at which the impedance curve shows the first minimum after the resonance frequency at increasing frequencies. It can be read directly from the impedance curve with the aid of the graphics cursor. In the program it is obtained by scrolling up through the impedance data array, starting at $f_r$; the minimum is considered found if N RUNAWAYS + 1 (see section 5.1.2, Special Notes on Entry of Parameters) are found in successive values in the increasing direction.

### 5.1.8 Q Factor

The quality factor $Q_t$ is derived from the impedance curve according to the following formula (BASIC syntax):

$$\boxed{Qt=Fr/(F2-F1)/R0*SQR(R0^2-R1^2)/SQR(R1^2-1)}$$

where

$Z_r$, $F_r$ = Resonance impedance and resonance frequency (see section 5.1.5)
$Z_t$, $F_t$ = Tuning impedance and tuning frequency (see section 5.1.7)
$R0$ = Resonance impedance standardized to $R_{DC}$ (see section 5.1.3, DC Impedance)
$R0=Z_r/R_{DC}$
$R1$ = Impedance standardized to $R_{DC}$ between $Z_r$ and $Z_{out}$ (see section 5.1.7, Tuning Frequency);
$R1=SQR(R0)$
$F1$ = Frequency below the resonance frequency at which impedance $Z1=R1R_{DC}$
$F2$ = Frequency above the resonance frequency at which impedance $Z2=R1R_{DC}$
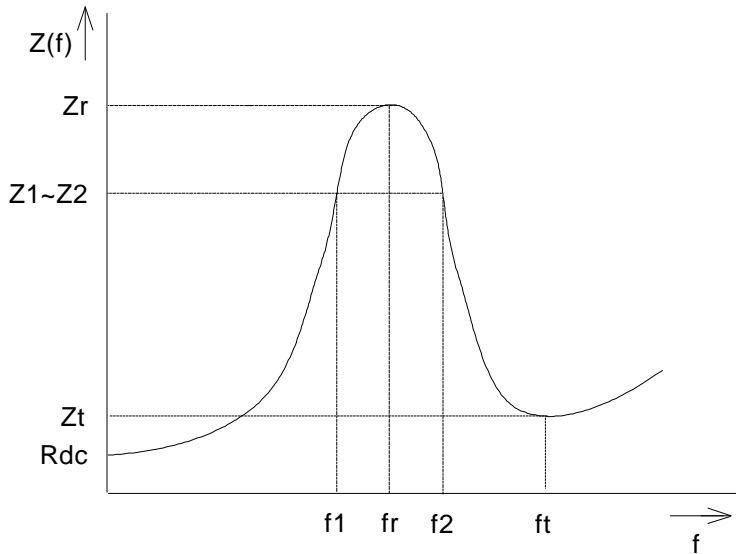


FIG 8: Parameters of impedance curve for determining Qt

### 5.1.9 Equivalent Volume

The equivalent volume $V_{eq}$ of a loudspeaker array defines the air volume, the compliance of which is equal to that of the loudspeaker array.

First the impedance measurement is carried out with the loudspeaker array open. The loudspeaker is then accommodated in a small, rigid housing without any further openings and additional insulation and a new measurement is started with the $V_{eq}/V_b$ softkey (F11, level 4). With the newly determined resonance frequency $F_b$ ($>F_r$) the equivalent volume $V_{eq}/V_b$ referred to $V_b$ is obtained with the formula

$$\frac{V_{eq}}{V_b} = \left(\frac{f_b}{f_r}\right)^2 - 1$$

$V_b$ being the internal volume of the loudspeaker housing less the volume of the loudspeaker array.

### 5.1.10 Go/Nogo Test

At the end of a sweep the impedance characteristic of a reference loudspeaker can be stored as a reference trace (softkey STO REF) and is then available as a "golden unit" for further measurements. To measure further DUTS, the stored reference characteristic is loaded with softkey REF SPK and the tolerance range specified. One of three display modes can be selected:

- TRACE A: the relative impedance characteristic is displayed.
- TRACE B: the relative impedance characteristic is displayed as trace B while the absolute impedance characteristic is displayed as trace A (see Fig. 7).
- OFF switches the reference off so that only the absolute impedance characteristic is displayed.

## 5.2 Sound Pressure Measurements

Sound pressure variations versus frequency allow conclusions to be drawn on the tone quality of a loudspeaker. Without this information dimensioning of crossover units is not possible. Furthermore, the sound pressure characteristic of a system provides information on possible polarity reversals in the loudspeaker system provided the sound pressure curve of a correctly poled reference system is available (see section 5.4.3, Polarity Check by Comparison to Reference Sound Pressure Characteristic).

With sound pressure measurements at a fixed frequency, the maximum sound pressure of a loudspeaker (at nominal power) and the sensitivity (at 1 W) can be measured at a distance of 1 m. The measured sensitivity (in dBspl at 1 W and 1 m distance from the microphone) may be stored as a reference value in the measurement of frequency response (see section 5.2.5, Sensitivity Measurement and Radiation Angle).

Sound pressure measurements (measurement of sensitivity and sound-pressure frequency response) are selected by means of the SOUND softkey (F7).

## 5.2.1  Test Setup

Required are a power amplifier, a calibrated standard microphone and possibly a microphone amplifier.
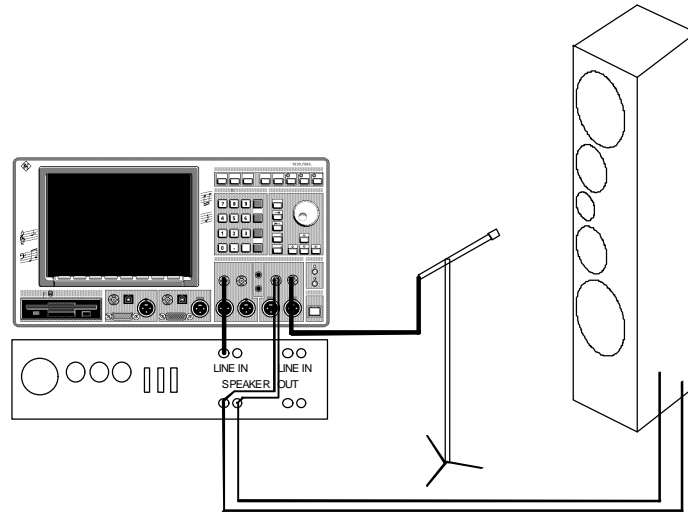


FIG 9: Test setup for measuring the sound pressure

The input of the power amplifier is connected to the XLR output (channel 1) of the UPD/UPL generator, the power amplifier output in parallel to the DUT and to the XLR input (channel 1) of the UPD/UPL analyzer.  This allows the output power of the power amplifier to be determined. For measuring the sound-pressure characteristic the standard microphone is connected to the XLR input (channel 2) of the UPD/UPL analyzer (if required via the microphone amplifier).

Sound measurements are preferably carried out in an anechoic chamber where the measurement cannot be falsified by indirect noise. Without the use of such a chamber, reflections caused by walls, floor or ceiling may amplify or attenuate the sound level at certain frequencies. The strongest amplification or attenuation occurs when the in-phase or quadrature signals are superimposed. This effect cannot be completely avoided but it can be reduced by means of the following measures:

- Covering all walls with absorbing material (carpets, curtains, etc) reduces the echo level.
- Minimizing the microphone distance. This increases the direct signal level in relation to the echo level.
- Maximizing the echo path by suitably positioning the standard microphone and the loudspeaker. This reduces the echo level and extends the period during which an echofree measurement can be made. In the ideal case and particularly at higher frequencies it can be attained that the measurement is terminated before the echo arrives at the microphone.
- Using noise or burst signals instead of continuous sinewave signals. Thus echoes of the previous measurement affecting the current measurement can be eliminated.

Sound pressure measurements can be carried out as far-field, near-field or combined near-field/far-field measurements.

## 5.2.1.1 Near-Field Measurements

For a near-field measurement the microphone is positioned as closely as possible to the loudspeaker. Thus an extremely high signal level is obtained and reflections are negligible in most cases.
Microphone and loudspeaker should be placed in the center of an imaginary sphere, the surface of which touches the nearest wall.
Near-field measurements can also be performed in rooms that are not completely sound-absorbing. However, the results obtained are only valid at low frequencies. The equation

$$2 \cdot \boldsymbol{p} \cdot r < \boldsymbol{l} \, , \qquad \boldsymbol{l} = c / f$$

yields a frequency limit of

$$f_{max} = \frac{c}{2 \cdot \boldsymbol{p} \cdot r}$$

with r being the radius of the loudspeaker.

## 5.2.1.2 Far-Field Measurement

For far-field measurements, the distance between microphone and loudspeaker should be at least 3-times the loudspeaker size. In practice, this distance is typically 1 m.
Microphone and loudspeaker should be placed at the focuses of an ellipsoid, the surface of which touches the nearest wall.
Since the far-field measurement is very sensitive to echoes it should be performed either in the open field or in an anechoic room. A time-selective measurement with burst signals is also recommended.

## 5.2.2 Adjusting the Output Power

The output power is adjusted by means of the POWERAMP softkey (F11, level 3) which opens a softkey menu where the user can select manual or automatic control and make necessary entries.

In both cases the nominal loudspeaker impedance has to be entered (REF IMPE, F6). The selection between the manual and automatic mode is made by pressing softkey MANU (F11) or AUTO (F12) which starts the measurement or control. After the start a new softkey level is displayed which offers only the STOP softkey (F10) for terminating the measurement.

Manual mode: After starting the power measurement the user has to adjust the amplifier exactly to the desired power. In this case the UPD generator outputs the constant voltage selected under VOLTAGE (F6, level 3). The amplifier output power is constantly measured by the UPD analyzer and displayed in a bargraph (trace B) and as a numerical value (function window of channel 1).

Automatic mode: The desired output voltage is adjusted through control of the generator output voltage. The user has to enter the desired power (POWER softkey, F8) and the maximum permissible amplifier input voltage (MAX VOLT, F9, for protecting the power amplifier). After the start (AUTO, F12) the generator output voltage is regulated (starting at 1 mV) using the specified values until the STOP softkey is pressed. The current output power (measured value) can be read in the display or seen from the bargraph. The power amplifier can be set before the start of the measurement routine and/or during the execution.

**Recommended:** *First set the power amplifier with MANU to roughly the required power (preferably to a slightly higher value) and then trigger the fine adjustment of the generator voltage with AUTO.*

*Note*:
1.   *After 10 measurements the control routine is terminated even if the STOP softkey is not pressed. This also applies when the desired output power cannot be reached with the generator voltage defined as MAX VOLT.*
2.   *Upon termination of the control routine, the current generator voltage becomes the new default voltage which is stored together with the other settings of the POWERAMP softkey menu when the application is quit.*

Selecting the POWERAMP menu implies that the power measured in the subsequent main measurement (response or sensitivity) is displayed on trace B as a curve or bargraph. If this is not desired (eg because the 2nd scale at the right-hand side of the curve or the 2nd curve are disturbing), trace B is to be switched OFF again

- prior to the first main measurement by briefly returning to level 2 and reselecting the measurement mode (not recommended in the FFT noise mode because of the noise generation time) or
- after the first main measurement by changing to the UPD display panel (see section 7.1, Change of Task Application/User Interface) and manual switch-off (OFF) of trace B.

## 5.2.3  Entry of Microphone and Room Parameters

The MICROPHN or MIC&ROOM softkey (F10, level 3) opens a softkey menu (level 4) for entering microphone and (in the case of sweep measurements) room parameters such as:

- sensitivity of standard microphone in mV/Pa (SENSITIV, F6);
- gain factor of microphone amplifier in dB (GAIN, F7), with no microphone amplifier connected a GAIN of 0 dB must be set;
- distance between standard microphone and loudspeaker in m (DISTANCE, F8) so that the sound travel time can be taken into account.

For sweep measurements (FREQ SWP or BRST SWP) the room parameters are entered in addition:

- Shortest echo path in meters (ECHOPATH, F9). Loudspeaker and standard microphone should be set up (in not completely sound-absorbing rooms) so that the echo path is as long as possible (see 5.2.1, Test Setup).
- Reverberation time in milliseconds (REVERBER, F10). A new burst measurement is delayed by this time to exclude echoes from the previous measurements.

The reverberation time can be determined using the WAVEFORM function. The generator is set to sine burst with a few periods of switch-on time and a long measurement interval (eg 60 s). A (single) waveform measurement reveals the burst periods and the decay of the reverberation. With the aid of the graphics cursor and the *-o function, the time between the end of the burst and the disappearance of the reverberation can be directly read.

When the sound pressure unit is selected (see section 5.2.4) the reference values for RMS measurement and display scaling are obtained from the entered microphone parameters.

*Note*: *Microphone parameters are stored when the application is quit (SPEAKER.CFG) and reloaded on the next call up. The default values for common 1/2" microphones such as B&K or Microtech Gefell are 50 mV/Pa, 0 dB (without additional microphone gain) and 1-m measurement distance.*

### 5.2.4  Selecting the Sound Pressure Unit

With the aid of the UNITS softkey (F9, level 3) the unit of the sound pressure displayed as a measured value or curve can be selected. This selection also determines the reference value and the reference mode (lin. in V/Vr or log. in dBr). The following units may be selected:

linear reference:
*   µBar (F6):          reference = sensitivity+gain-20dB;
*   Pascal (F7):          reference = sensitivity+gain;

logarithmic reference:
*   dBspl (F8):          reference = sensitivity+gain-94dB;
          0 dBspl (sound pressure level) is defined as 0.0002 µBar (= 20 µPascal);
          20 * lg (0.00002) dB = -93.98 dB
*   SENSITIV (F9)       reference = stored loudspeaker sensitivity
          (see 5.2.5, Sensitivity Measurement and Radiation Angle)
*   dBr (F10)          reference in dBV (may be entered);
*   REF SPK (F11)       reference is the frequency response of a "golden unit".
          Thus the GO/NOGO mode is implicitly switched on and the file name and tolerance values are entered.

The sensitivity is that of the standard microphone in dBV/Pa and gain that of the measuring amplifier (see section 5.2.3, Entry of Microphone Parameters).

Entry in dBr (F10) is useful, for instance, when the sensitivity of the sound pressure meter is specified as

$$\Rightarrow \qquad 1 \text{ V} \; \overset{\wedge}{=} \; 0 \text{ dBV at 94 dBspl} \qquad ==> \qquad \text{reference = -94 dBV}$$

or if a specific voltage generated by the microphone should be stored as a 0-dB line.

***Note****: In the case of sensitivity measurements the units µBar and Pascal cannot be entered. If they are already set (via a response measurement), they are automatically changed to dBspl.*

### 5.2.5  Sensitivity Measurement and Radiation Angle

The sensitivity measurement is a sound pressure measurement at a fixed frequency (measurement mode SNS SNGL) or in a defined frequency band (measurement mode SNS RAND).

After selecting the measurement mode, the measurement frequency or the frequency band can be entered in addition to the entries mentioned above (see sections 5.2.2 to 5.2.4) and to the selection of the generator voltage. (SNS FREQ or SNS RNG). The entered frequency also serves a reference frequency for the 0-dB line in the measurement of the transfer function. If no entries are made at all, the entries of the last sound pressure measurement will be used.

Normally, sensitivity is specified in dBspl at 1 W and 1 m. For this reason the power amplifier should first be set to 1 W (see section 5.2.2, Setting the Output Power) and the unit dBspl be selected (see section 5.2.4, Selecting the Sound Pressure Unit).

Pressing the START softkeys triggers a continuous level measurement (without graphics display of measured values).  Results are displayed in V/Vr or dBr and should be interpreted with the unit selected under section 5.2.4 (Selection of Sound Pressure Unit).

The measured sensitivity may be used as a reference for measuring the frequency response or the radiation angle. This is done by pressing the STO SENS softkey (F10).

This measurement can also be used for determining the radiation angle. In this case the standard microphone (at a constant distance from the loudspeaker) has to be moved away horizontally from the reference axis in the course of the long-term level measurement until the sound pressure level falls by 10 dB. If the -10 dB should be directly displayed, the sound level unit SENSITIV has to be selected (after the measured value has been stored at 0°).

### 5.2.6 Frequency Response of Sound Pressure Level and Transmission Range

Three different test methods can be selected for measuring the sound pressure frequency response:
- FFT RAND uses a special noise signal for fast measurements of the entire frequency range.
- FREQ SWP is a frequency sweep with sinewave signals for measurements under open-field conditions (eg in an anechoic chamber) and near-field measurements.
- BRST SWP is a frequency sweep using burst signals for far-field measurements (also in not completely sound-absorbing rooms).

Although the **FFT noise** measurement ensures a fast *display* of the transmission range, it can only be used with restrictions as far as numeric *calculations* are concerned:

- If a *narrow FFT* is selected (1 or 2 k FFT), a coarse frequency resolution is obtained and little information particularly at low frequencies.
- With *large FFTs*, the measurement time itself is still very short but much more samples have to be measured for determining the transmission range than with a sweep so that the speed advantage of the FFT noise measurement diminishes with increasing FFT size.
- When *frequency limits are changed,* the noise signal has to be recalculated so that a deadtime from several seconds to one minute may occur, depending on the selected FFT size and frequency bandwidth.
- When *the FFT size is changed,* also the sensitivity has to be recalculated.
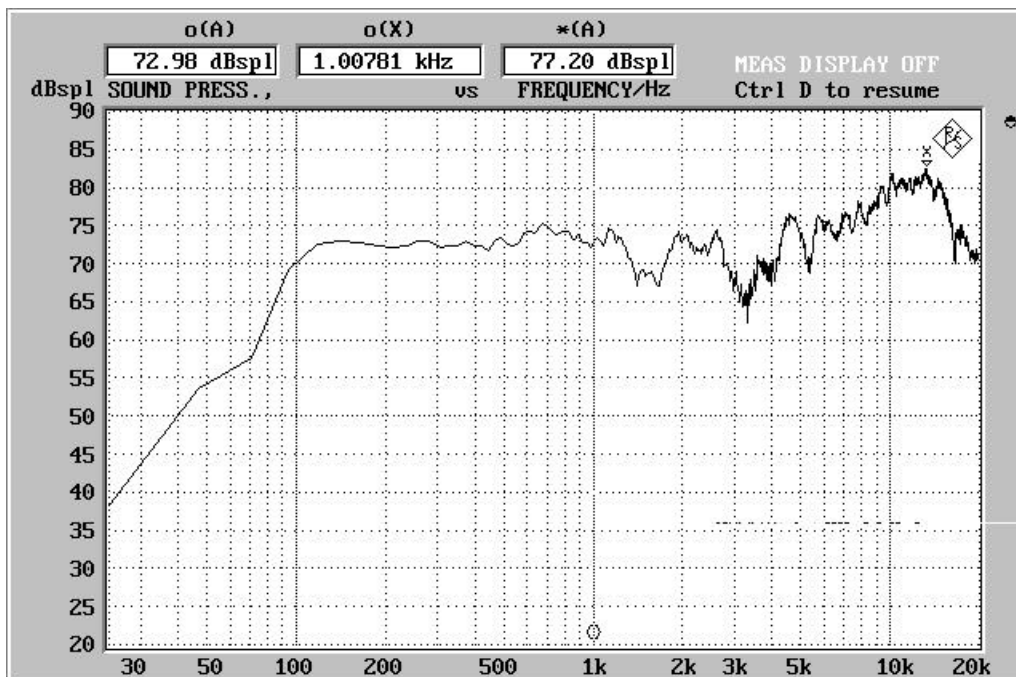


FIG 10: Measuring the sound pressure level using FFT noise

Since the **burst sweep** is not directly available as a measurement function in the UPD or UPL, its function will be briefly described below:

Burst sinewave signals with a length of integral periods of the sweep frequency are used as measurement signals (to avoid DC components). The number of burst periods is selected such that no more periods follow after the end of the measurement. The measurement is performed with a special RMS mode (TRIGGERED). After the start of the measurement, the mode waits for the first burst signal to arrive (ie a selected trigger threshold to be exceeded) and then starts the RMS measurement.

The measurement time has to be selected so that

- integral multiples of a period (at least one) are measured to avoid errors caused by interruptions,
- a maximum number of periods is measured to obtain a high accuracy,
- measurements are only performed before the first echo arrives.

Echofree measurements can only be performed in the short time between the echo delay and the signal delay. The frequency of exactly one measurement period during this time is the theoretical lower frequency limit.

$$f_{\min} = \frac{1}{T_e - T_a} = \frac{330m/s}{L_e - L_a}$$

where $L_a$ = microphone distance, $L_e$ = shortest echo path.

This means that below this frequency limit also parts of the echo signal are measured. In spite of this, these measurements still have an advantage as against the sinewave sweep. Since the signal is switched off after the measurement and a new measurement is only started after the elapse of the reverberation (REVERBER, F10 under MIC&ROOM), the measurements cannot be affected by echoes from a previous measurement. In practice, the frequency limit is slightly higher because of the delayed start of the measurement.
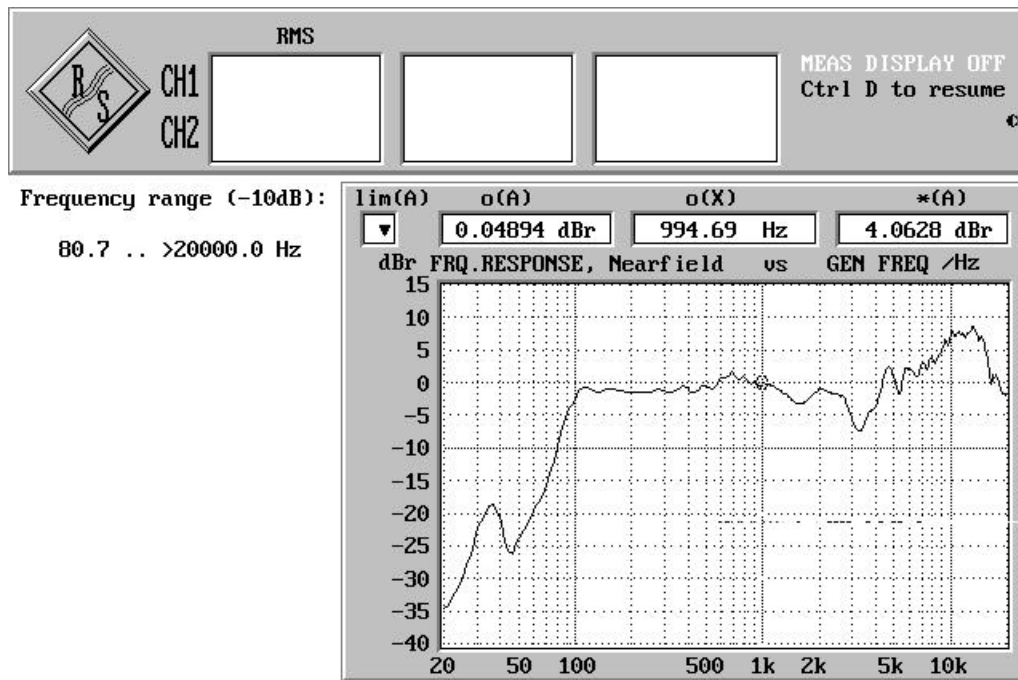


FIG 11: Measurement of transmission range using a burst (far field) and a sinewave sweep (near field).

The **sinewave sweep** should be used whenever no significant echoes are expected, eg in the near field or in anechoic chambers. Near-field measurements in the combined near/far-field measurement are always performed with a sinewave sweep.
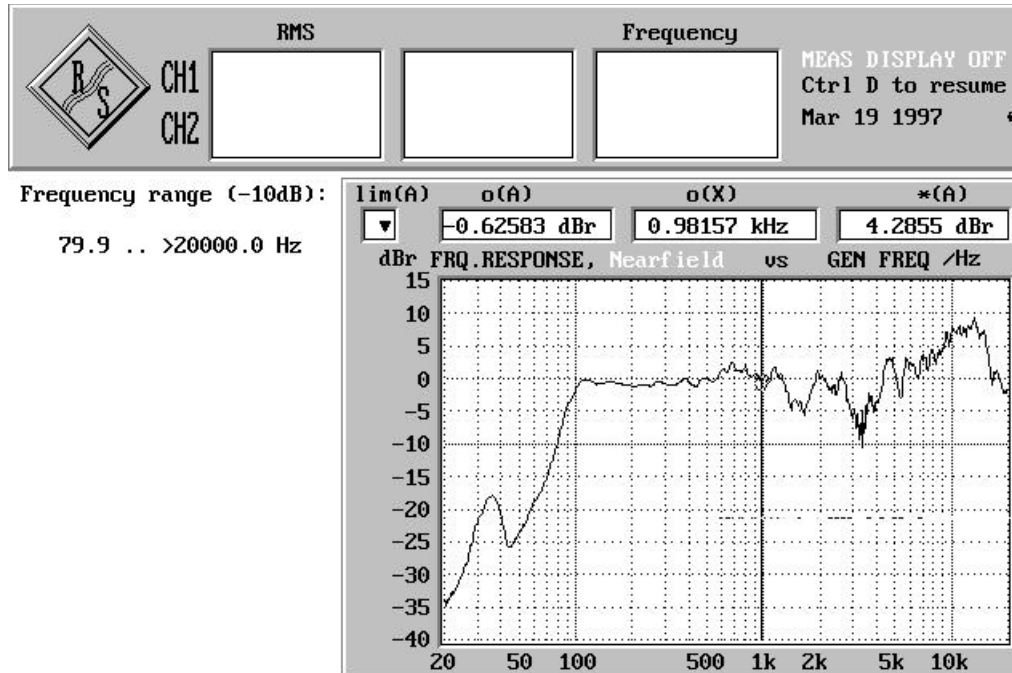


FIG 12: Measurement of transmission range using a sinewave sweep in an anechoic chamber

With the START softkey a level sweep or an FFT is started; the graphics display is in the unit selected under 5.2.4, Selecting the Sound Pressure Unit.

SENSITIV has to be selected for an automatic determination of the transmission range after the measurement. Full-screen display must be deactivated for the output to have enough space on the screen. After the display of the frequency response characteristic the transmission range is calculated, defined by a lower limit of -10 dB. When this limit is exceeded for more than 1/3 third (1/9 octave), the limit of the transmission range is attained. Shorter drops are ignored. The transmission range determined by the application program is indicated next to the graphics display.

Determination of the transmission range is started at the frequency specified under SNS FREQ. If the level at this frequency is below -10 dBr, the calculation is stopped and a respective error message output.

## 5.2.7 Selecting the Combined Near/Far-Field Measurement

For the two sweep measurements, the far-field and the near-field can be configured separately, determined with different test setups and frequency response curves can be combined.

The frequency range and the number of sweep points for the far field are determined under CFG FAR (F7). If an additional near-field measurement is not performed, the complete frequency response measurement is carried out with these settings.

The frequency range and the number of sweep points for the near field are determined under CFG NEAR (F8). If an additional far-field measurement is not performed, it can be switched off here (F6, the ON or OFF state is indicated in the softkey). The transition frequency between the near and the far field (TRANSFRQ, F9) can be preset as a numerical value. A sinewave sweep is always performed for the near-field measurement.

If an additional near-field measurement should be carried out, the near and the far field should sufficiently overlap and the optimum transition frequency be graphically determined at the end of the measurement.

With the near-field measurement ON (F6 under CFG NEAR), the question

⇒      "Continue with near field measurement? - [y]/n".

is displayed after the (far-field) sweep. The test setup can now be modified, ie the microphone may be positioned in the near field. The near-field sweep can be started by striking any key (except N or n). During the near-field measurement, the two measurements are graphically displayed (the far-field measurement as trace B). The graphics cursor is on the preset transition frequency and can be shifted at the end of the measurement by means of the left and right cursor keys. By striking any key, the actual transition frequency is confirmed and the two curves are combined; the near-field measurement is rescaled to the level value of the far-field measurement at the transition frequency.

***Note****:*
· *The two curves are still available as separate data records after they have been combined. They can be combined again by entering a new TRANSFRQ (F11 in level 4).*
· *A combined near/far-field measurement is not possible for the GO/NOGO test (selected by UNIT REF SPK, see 5.2.4, Selecting the Sound Pressure Unit).*
· *As echoes normally do not occur in the near field, the near-field measurement is carried out as a normal sinewave sweep even if BRST SWP is selected.*


## 5.3  Phase and Associated Parameters

The demands for phase linearity of audio components have gained importance in the past years. This also applies to loudspeakers where some manufacturers avoid phase shifts by using single systems - without crossover units.

Normally, phase nonlinearities are detected in a *linear* frequency sweep (see section 5.3.2, Frequency Response of Phase). The phase delay at a fixed frequency also provides information on the polarity of the loudspeaker array or system (see section 5.4.2, Polarity Check by Comparison to Reference Phase Measurement).

Phase measurements are selected with the PHASE softkey (F8). At the next softkey level (2) a single measurement (SINGLE, F6), eg for polarity checks, or a (linear) sweep (SWEEP, F7) can be selected.

### 5.3.1  Test Setup

The test setup is the same as for sound pressure measurements (see section 5.2.1), but the use of a power amplifier is not obligatory. In a setup without power amplifier, the generator outputs are  connected in parallel to the loudspeaker and to the XLR input of the UPD/UPL analyzer (channel 1).
The standard microphone is connected to the XLR input (channel 2) of the UPD analyzer (if required via a microphone amplifier).
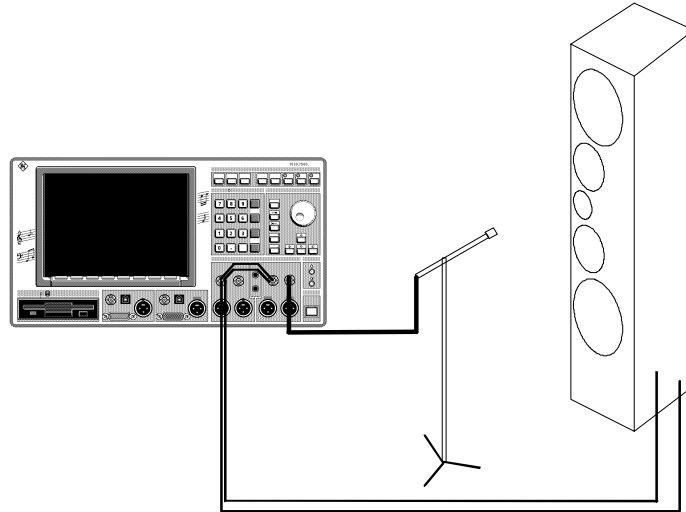


FIG 13: Test setup for phase measurement

### 5.3.2  Frequency Response of Phase

The phase linearity should be checked in a linear frequency sweep where the phase is measured between the loudspeaker input and the generated sound wave. In the case of a linear phase response, a line with a negative slope would ideally be obtained. In order to obtain a continuous phase response without 360° shifts, the sweep points displayed in the range from 0 to 360° have to be further processed.  The algorithm of the application software requires in addition that the phase values measured at two adjacent sweep points differ by less 180°. At a defined microphone distance the signal travel time is

$$t\,/\,s = \frac{d\,/\,m}{330}$$

The requirement

$$\boldsymbol{j}_2 - \boldsymbol{j}_1 = 360° \cdot (f_2 - f_1) \cdot t < 180°$$

yields for the frequency increment

$$(f_2 - f_1)\,/\,Hz < \frac{1}{2 \cdot t\,/\,s}$$

or after the start of the signal travel time

$$(f_2 - f_1) / Hz < \frac{165}{d / m}$$

Same as with sound pressure measurements, the microphone distance is entered with the DISTANCE softkey (F8, level 4) in the MICROPHN menu. The distance entered in this menu is used as a default value (see DISTANCE in section 5.2.3, Entry of Microphone Parameters). The program calculates the sweep steps from the distance of the microphone according to the formula

$$(f_2 - f_1) / Hz = \frac{150}{d / m}$$

Note that the microphone should not be set up too far away as otherwise the desired frequency range cannot be covered with the maximum permissible sweep points (1024). The condition

$$\left| f_{stop} - f_{start} \right| / Hz < 1023 \cdot \frac{150}{d / m}$$

should be fulfilled, or for the microphone distance

$$d / m < \frac{153450}{\left| f_{stop} - f_{start} \right| / Hz}$$

***Example****: A microphone distance of 1 m and a frequency sweep of 20 to 20000 Hz yield a frequency increment of 150 Hz and therefore 135 sweep points. As the example shows, this restriction has relevance only in theory.*

If the sweep should be carried out with a greater number of points than required by the microphone distance, this can be obtained by entering SWP PNTS (F8, level 3, see section 3.4, Entry of Parameters).

The display mode can be selected either before the sweep is started or after its termination (see section 5.3.3, Linearity Departure and Group Delay).

***Note****: In the case of phase measurements, only DISTANCE is offered in the MICROPHN menu; all other microphone parameters are irrelevant.*

### 5.3.3  Linearity Departure and Group Delay

Assessing the phase linearity from the sweep diagram is problematic because a range of up to 180000° (1024 * 180°) has to be displayed in the Y direction with linear scaling. A better overview is provided by two alternative display modes, linearity departure and group delay. They are selected with softkeys F9 to F11 at softkey level 3 or 4 before or after the sweep.

- PHASE (F9) displays the sweep results unchanged with absolute phase values,
- LINEAR (F10) displays the deviations from an ideal straight line. The slope of this line is determined between 2 frequencies to be entered in REF AREA.
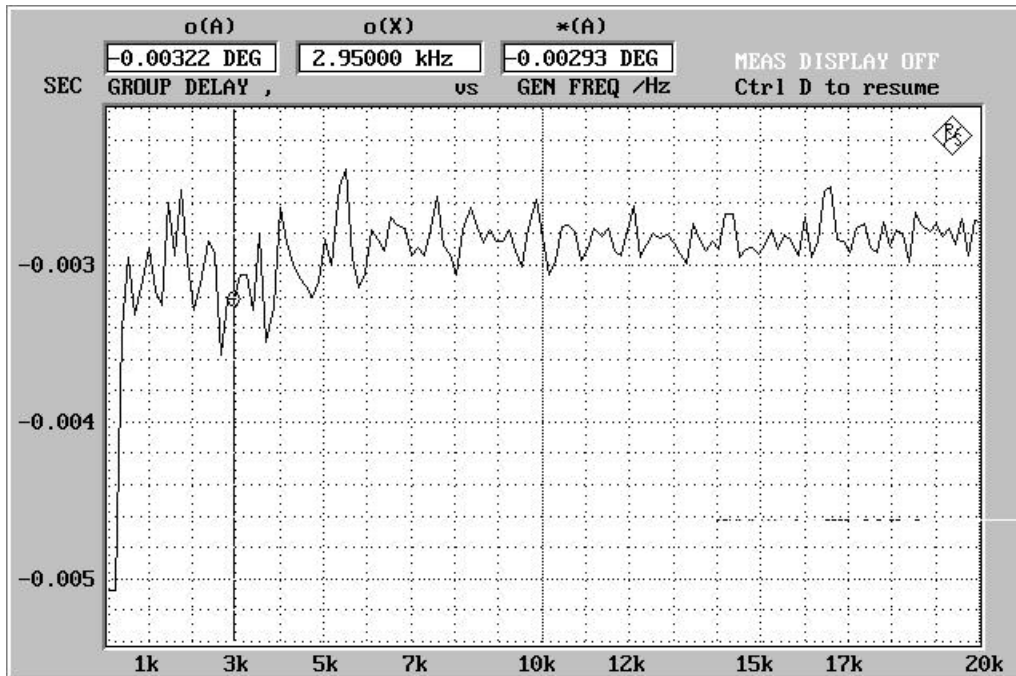- GRP DEL (F11) displays the group delay versus frequency.



FIG 14  Group delay derived from the phase characteristic

## 5.4  Polarity

Correct polarity of crossover units, loudspeaker arrays and systems becomes important if partly or completely identical signals are applied to several loudspeakers. While an incorrectly poled mono broadband loudspeaker would hardly be noticed, superposition of the 180°-phase-shifted sound waves of two stereo loudspeakers produces audible cancellations. This also applies to incorrectly poled single systems in a multiple loudspeaker, where cancellations occur also in the transition range of the crossover unit.

### 5.4.1  Measurement Function POLARITY for Electrical Systems

UPD and UPL provide a measurement function and a corresponding generator function for measuring the polarity of electrical systems (cables, crossover units). In this case a (DC-loaded) $\sin^2$ pulse is generated and the measured polarity is displayed as "+ 1 Pol" (correct polarization) or "- 1 Pol" (wrong polarization).

However, this method cannot be used for loudspeakers.

### 5.4.2 Polarity Check by Comparison to Reference Phase Measurement

This procedure is based on the fact that incorrect polarity of a system causes a phase shift of 180°. For the check a correctly poled reference system is set up at an exactly defined distance from the standard microphone and the phase between generator signal and output of the standard microphone (or microphone amplifier) is measured. This measured phase is stored as a reference value and an upper and lower limit value of $\pm 45\ \Delta°$ is defined.

With a reference value below 45°, phase measurement mode -180 to +180° is selected, in all other cases mode 0 to 360°. The DUT is then set up at the same distance and the phase is measured (at the same frequency). If the measured value is outside the selected tolerance, the loudspeaker is incorrectly poled.

The inaccuracy of the distance between loudspeaker and standard microphone must not exceed 1/8 of a wavelength (45/360). For this reason this method cannot be used at high frequencies. For the upper limit frequency $f_{lim}$ the following equation applies:

$$f_{\lim} = \frac{v}{l} \cdot \frac{45}{360} = 300\frac{m}{s} \cdot \frac{0.125}{l} = \frac{41.25 m/s}{l}$$

- v  being the sound velocity in m/s, and
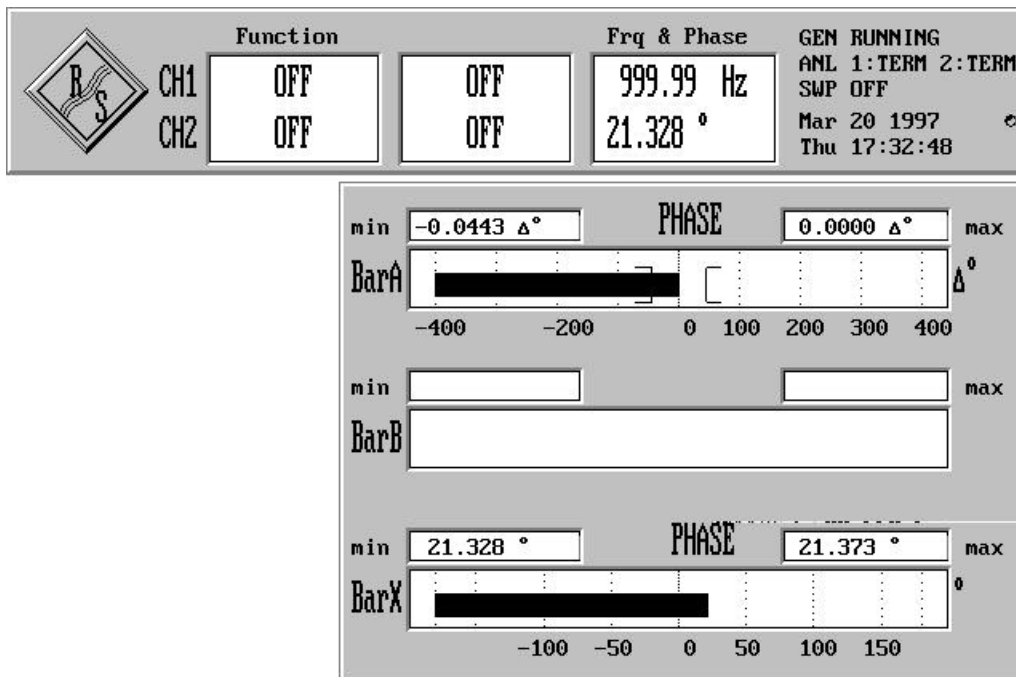- l  the maximum deviation of loudspeaker - microphone distance in m.



FIG 15 Polarity check by phase comparison with a reference loudspeaker of correct polarity
(a phase difference of approx. 0° means that the polarity is correct, the absolute phase has no significance)

This measurement can be selected under PHASE (F8, level 1) as SINGLE mode (F6, level 2).

### 5.4.3 Polarity Check by Comparison to Reference Sound Pressure Characteristic

This method uses the superposition of frequencies in the transition range of the crossover units. If one of the loudspeakers is wrongly poled, some frequency components are cancelled and this is clearly reflected by the frequency response of the sound pressure level (particularly when compared to that of the correctly poled reference loudspeaker).
This method should be preferred for checking the polarity of tweeters and mid-range speakers.

## 5.5 Distortion

The UPD and UPL device software offers several functions for determining the distortion of a signal - and thus also of the sound pressure produced by a loudspeaker. The following functions are available for example:

- THD for measuring the total harmonic distortion or any combination of discrete harmonics with or without display of the harmonics line spectrum. This measurement function ignores the noise; for measuring the 2nd and 3rd order distortion in line with DIN IEC 268, part 5, $d_2$ and $d_3$ have to be selected.
- THD+N with or without post-FFT of total signal,
- FFT with or without zoom FFT for increasing the frequency resolution.
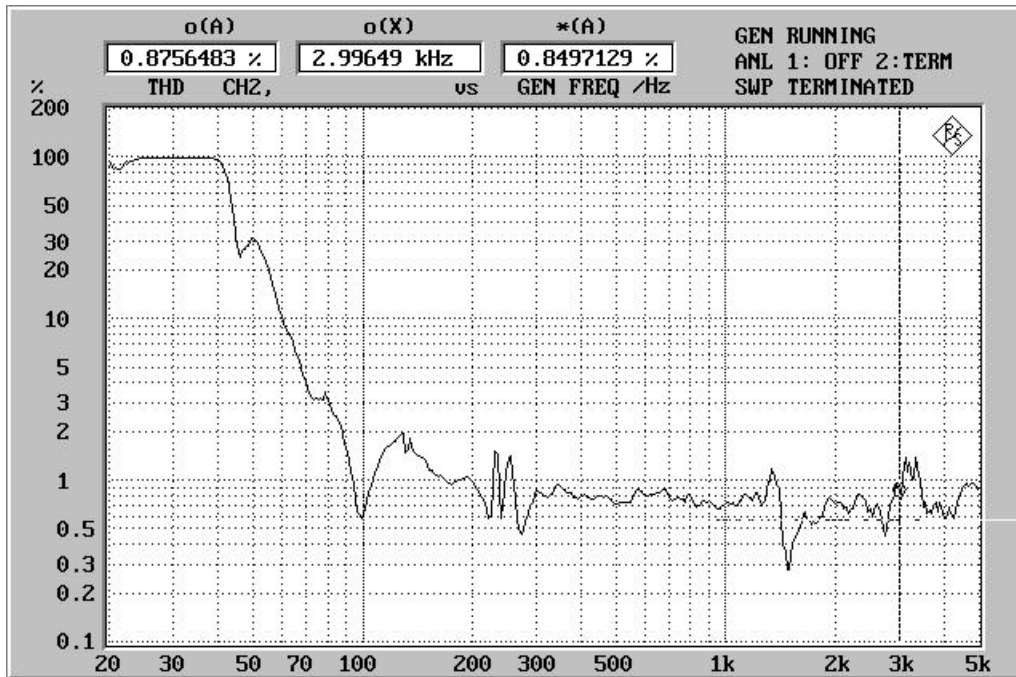


FIG 16 Distortion (THD) versus frequency

Although these measurement functions are directly accessible from the user interface, the most important are also integrated at the application level and may be selected with the THD softkey (F9) at the main level. At the next softkey levels different kinds of measurements can be selected:

- THD TOTL (F6)     total distortion (all di)
- THD d2 (F7)       2nd order distortion
- THD d3 (F8)       3rd order distortion
- THD+N (F9)        total distortion with noise
- POST FFT (F10)    THD+N with display of total signal as post-FFT
- ZOOM FFT (F11)    display of total signal with zoom factors from 2 to 128

Except for ZOOM FFT, measurements can be carried out as single measurements (SINGLE, started with F11) or in the form of a sweep (SWEEP, F12). After pressing the SNGL FRQ softkey (F10), the frequency for a single measurement is entered under FREQ RNG (F7) the frequency range for the sweep. % or dB can be selected under UNIT (F9).

For single measurements the measured values are also numerically displayed. During sweeps the result display is switched off as standard to increase the sweep speed. To switch on the result display a single measurement should be performed before the sweep is started.

Prior to each RESTART of a single measurement, the frequency of the useful signal can be changed at the post-processing level (SNGL FRQ, F10).

DIN IEC 268, part 5, recommends frequency sweeps up to 5 kHz; measurements at higher frequencies are not useful as the 22-kHz analyzer measures the 3rd order distortion for instance only for signals up to approx. 7 kHz.


### 5.5.1  Special Features of THD+N Measurement (with post-FFT)

The relatively high distortion factor of sound converters permits measurements to be made in the FAST mode. In this mode of the THD+N measurement useful-signal suppression to obtain precision is left out. This halves the measurement time which is particularly important for swept measurements.
Since the useful signal is displayed in its original size, post-FFT shows harmonics and useful signal in their correct proportion.

For measurements without post-FFT display, the FFT size for *calculating* the THD+N value is reduced from 8 to 1k to further increase the measurement speed.


## 5.5.2  Special Features of Zoom FFT

The FFT measurement function is offered exclusively as zoom-FFT as normal FFT is displayed (and swept) as post-FFT of the THD+N measurement.

FFT size and zoom factor can be set via the 1-out-of-N selection softkey, the zoom factor as a power-of-two integer from $2^1$ (2) to $2^7$ (128).

FFT display is normally in Volts while the distortion (also with post-FFT) is displayed in % or dB. In order to obtain the same scale for FFT and post-FFT, the FFT is processed after the first measurement:

- set the graphics cursor to maximum (also the useful signal),
- the voltage value found there is displayed and stored as a reference value for the display scale,
- the value is used as reference depending on the selected unit:
    - %V/Vr, if the selected UNIT is %,
    - dBr, if the selected UNIT is dB.

# 6  Combining Several Measurements

In many cases it may be required to start operating steps or whole test sequences by a single keystroke. If a certain test sequence has been tried out and is to be frequently used, it should not be necessary to set each measurement step manually. When the test program is used in production, automatic test sequences are an absolute necessity to avoid operating errors and to shorten the throughput time.

The tools required for combining measurements are available under the UTILITY softkey (softkey level 1).

## 6.1  Macro Operation

The simplest way to generate and perform automatic test sequences is the use of macros. A sequence of keystrokes is stored and repeated in the same order when required. Macro operation in a way simulates an operator performing the same operating steps over and over again.

To *create* a macro, the keystrokes are written into a file, which is closed after recording, and stored under a reserved (MACRO #0 to MACRO #5) or a user-defined name (USER MAC).

To *call up* a macro, these files are loaded by a keystroke (or by entering the file name in the case of USER MAC). All stored keystrokes are sequentially simulated until no more information on keys is left in the file or (in the case of a GO/NOGO test) a "DUT failed" occurs.

To ensure proper function of the procedure, recording and calling up must be started from the same point. This is ensured by the application software where the common start point is stored in the menu under UTILITY -> MACRO.

Any number of macros can be defined. 7 sample macros (SPEAKER0 to SPEAKER6)  have been defined and assigned to the softkeys MACRO #0 to MACRO #5. They can be called up via USER MAC and may be overwritten any time. Macros can be prepared and stored under any name with the aid of the USER MAC softkey.

### 6.1.1  Recording a Macro

To record a macro press the STRT REC softkey and enter the memory location of the macro (MACRO #0 to MACRO #5) or a file name (USER MAC). The recording mode is indicated by the MACRO REC label in the display. Press the required softkeys as in normal operation. The keystrokes for several measurements can be performed in succession so that also the measurements will be sequential. Recording of a macro can be terminated in two ways:

- by pressing the ESC key at any point in the program. The macro is terminated immediately and the user may continue manual operation (eg restart the current measurement, trigger a screen printout, etc).
- by returning to the start point via the softkey menu and then pressing the STOP REC softkey. This has the advantage that a new macro (or the same one) can be started right away.

In both cases the macro is stored under the preselected name or softkey when recording is terminated.

***Note***:

· *If parameters are to be changed while a macro is performed (eg between two measurements), a new configuration file should be loaded (see 6.2).*

· *If the BACK key is pressed immediately after the STRT REC softkey, recording is not started as no memory location has been defined for the macro. The application is in the normal operating mode, ie MACRO REC is not displayed.*

### 6.1.2 Running a Macro

A macro is started by one of the START softkeys in the macro menu.

- **VIEW RUN** starts the macro and waits for a (predefined or explicit) macro name to be entered. The sequence is then performed as in manual operation using softkeys.
- **FAST RUN** has the same function as VIEW RUN with the exception that no softkey labels are displayed. This speeds up the execution of the macro.
- **RE-RUN** restarts the last macro without the macro name having to be entered. The selected mode (VIEW or FAST, possibly also "single step", see below) remains unchanged.

The user may switch on and off the single step mode for running a macro by specifying **STEP MOD.** A macro started in the single step mode expects a character to be entered after each read and executed softkey. This mode is used to check the macro function.

Macro operation is signalled by the message MACRO RUN. If a "DUT failed" occurs, the macro stops and waits for a keystroke. If a general error occurs the macro is aborted.


### 6.1.3 Predefined Macros

The six softkeys MACRO #0 (F6) to MACRO #5 (F11) are assigned the following functions:

MACRO #0: Go/Nogo test: Entry and automatic setting of output power (into 4$\Omega$), frequency response measurement of a reference loudspeaker by means of a sinewave sweep with display of sound pressure characteristic (in dBspl) and output power (in W), storing the sound pressure characteristic of the reference loudspeaker, switchover from UNIT to REF SPK with entry of upper and lower limit value, measurement of a test loudspeaker and check for compliance with tolerances, printout, termination of the macro.

MACRO #1: Frequency response measurement by means of FFT noise with automatic setting of a 1-W output power (into 4 $\Omega$) and determining the sensitivity in the frequency band from 500 Hz to 1.5 kHz.

MACRO #2: Frequency response measurement by means of a burst sweep. A near-field measurement is optional.

MACRO #3: Sweep of THD measurement (d2 to d9) with trace display.

MACRO #4: Sweep of THD+N measurement with display of post-FFT for each test point.

MACRO #5: Phase measurement between amplifier output and microphone input, printout, phase response measurement by means of sinewave sweep, display of absolute phase, linearity departure and group delay.

A 7th macro for an impedance measurement with analysis can be started by pressing the USER MAC softkey and entering the file name SPEAKER6.MAC:
- Display of impedance characteristic with computation of Thiele-Small parameters of a "golden unit" and storing the characteristic as a reference trace.
- Measurement of an additional loudspeaker and display of deviations from the reference loudspeaker.
- Measurement of an additional loudspeaker and display of impedance characteristic and deviation from the reference loudspeaker. This measurement is printed out.
- Switchover to the FFT noise mode, display of impedance characteristic and computation of Thiele-Small parameters.
- Termination of the macro.

Each of these macros first loads the associated configuration file (SPEAKERx.CFG, x=0..6, see 6.2). Exception: SPEAKER6.MAC calls up USER CFG permitting the entry of different configuration file names. At the end of each measurement the result is printed out if a configured printer is connected. (In the default state printout is to the PCX file SPEAKER.PCX).

*Note*: *Upon installation of the SPEAKER software the installation program checks whether the macros SPEAKER0.MAC to SPEAKER6.MAC do already exist. If this is the case, the macros are not re-installed to prevent user-defined macros to be overwritten. If a re-installation is desired, the macros in the SPEAKER directory must first be cleared. Command line (at the operating system level):*

    ⇒      DEL  C:\SPEAKER\SPEAKER?.MAC

## 6.2  Creating and Using Different Configuration Files

All settings required for the application are stored in the default configuration file SPEAKER.CFG and loaded when the application is started. It may be desirable to overwrite this data record with other settings during the application run, eg when another type of DUT is to be tested. In this case it is not necessary to call up menu items and enter new parameters. These presettings are particularly important in macro operation where the manual entry of parameters is rather tiresome.

The *current configuration is stored* with the softkey sequence

    ⇒      UTILITY -> CONFIGUR -> STORE -> xxxx,

where xxxx is either

- a predefined memory location CONFIG 0 to CONFIG 5 or
- an explicit file name (via USER CFG).

A *configuration is loaded* - ie the current configuration is overwritten - by pressing softkeys

    ⇒      UTILITY -> CONFIGUR -> LOAD -> xxxx,

where xxxx has the same function as for storing.

*Note*:
·  *With USER CFG, a macro can be used with different configuration files.*
·  *Loading a configuration overwrites the currently used configuration. If the currently used configuration should be saved, it must first be stored in one of the configuration files and then called up again.*
·  *If a configuration file has been loaded by mistake, which should not become the new default configuration, the application should be quit with CTRL C.*
·  *The installed configuration files should be used as standard with the macros of the same name. They should however be adapted by the user to the DUT. To do so each configuration has to be loaded, modified by entering the desired parameters (via the softkey tree or by entering values) and stored again.*
·  *When the SPEAKER software is installed, a check is made during the installation whether the configuration files SPEAKER0.CFG to SPEAKER5.CFG already exist. If this is the case they will not be re-installed to prevent user-defined files being overwritten. If a re-installation is desired, the configuration files in the SPEAKER directory must first be cleared. Command line (at the operating system level):*

    ⇒      DEL  C:\SPEAKER\SPEAKER?.CFG

## 6.3  Labelling of Macro and Configuration Memory Locations

To replace the neutral designations of the macro and configuration memory locations by user-defined ones, the BASIC lines 10110 or 10180 in the UTILITY.BAS module have to be changed. Eight characters are available for each softkey; the total length of the string must not be changed.

# 7 Processing of Measurements

In most cases the results need to be printed at the and of a measurement and changed to a specific format. To be able to fulfil all requirements, the application software allows modifications (eg of the graphics display) to be made at the user interface of the device software, then to return to the universal sequence controller and continue the program run. There are several alternatives for printing out the results.
These features are offered at softkey level 4, which is automatically selected at the end of a measurement.

## 7.1 Change of Task Application/User Interface

At the end of each measurement the GOTO UPD softkey (F7) is displayed. This allows a temporary change to the UPD or UPL user interface eg for

- changing the scale of the graphics display,
- selecting other measurements (frequency measurement, input-peak measurement),
- switching the 2nd trace on or off or changing it (eg to frequency display),
- switching the graphics cursor on or off or shifting it,
- adding a comment to the graphics display,
- adding a comment to the printout,
- reconfiguring the printer interface,
- printing out the graphics window together with a special status panel and comments,
- displaying limit violations in tabulated form (OPERATION  LIM REP).

Return to the application software is possible with the F3 key. After the return all application softkeys are ready for use again.

## 7.2 Printout

To print hardcopies, a printer has to be connected to the UPD/UPL and configured. It is important that a suitable printer driver is selected. Respective information is given in the UPD and UPL operating manuals. Upon delivery, the SPEAKER.SCO setup of the application software is configured so that the screen content is output to a SPEAKER.PCX file which is overwritten each time a hardcopy is made.

The screen content can be output (to a printer or file) either at the user interface or directly from the application software.

The following displays can be printed from the application:

- measured values of the application software together with the graphics sections,
- full graphics display with or without printout comment.

Printout of instrument panels (eg for the documentation of settings) is only possible from the user interface:

- 1 panel with graphics window, with or without printed comment
- 3 panels (without graphics), with or without printed comment

In addition, trace data and, if applicable, limit violations can be printed out (see section 7.2.1, Printout at User Interface).

### 7.2.1  Printout at User Interface

The application has to be quit by means of the GOTO UPD softkey (F7), (see section 7.1, Change of Task Application/User Interface). Same as with normal instrument operation, printout is triggered with the HCOPY hardkey. Any comment can be entered.

Command:

  $\Rightarrow$    PRINT-->Type

is provided in the OPTIONS panel for printing out trace data and limit violations,

- Type TRACE A (for trace data) or
- Type LIM REPORT (for limit violations)

for printout of an ASCII list with corresponding numerical values.

### 7.2.2  Printout During Running Application

At the end of each measurement softkeys HARDCOPY (F8) and o <--> O (F6) are offered. Thus it is possible to display or printout measured values with graphics window or full-screen graphics. Comments cannot be entered but will be printed provided the respective settings have been made in the OPTIONS panel.

## 7.3  Restart

### 7.3.1  Restarting a Measurement

At the end of each measurement a new measurement can be triggered by means of the START softkey (F12). Modifications made after the end of the first measurement at the user interface (eg special graphics scaling) are maintained until the program returns to the softkey level for measurement mode (or measurement) selection (see section 3.2, Softkey Levels).

### 7.3.2  Restarting the Application

The application can be started again without reloading by means of the RUN softkey (provided the application is properly concluded) or by entering the BASIC command RUN.

## 8 Terminating the Application

Normally, the application software is terminated with the EXIT softkey (F5) at the top softkey level (see section 3, Operation Concept). Since at lower softkey levels the left-most softkey (F5) is assigned the BACK function, the application can be terminated from any softkey level by pressing the F5 softkey several times. As a result all variables are cleared and the original softkey assignment and functions are restored. The configuration file SPEAKER.CFG is updated with the last user entries.

The application can be quit any time by means of the ESC key. To avoid inadvertent termination of the application, the query

&rArr;  "Exit this application? - y/[n]"

is displayed before the application is quit and has to be confirmed with Y. Any other entry causes the application to be continued.

The application can be aborted any time by means of the key combination CTRL BREAK. Softkeys and variables remain unchanged, likewise the configuration file SPEAKER.CFG. This avoids the  configuration file being overwritten. By entering

&rArr;  "CONT <CR>"

the program can be continued at this state.

***Note***: *After program abort (with CTRL BREAK or after an error has occurred) the original labels and functions of the softkeys can be restored with the BASIC command SOFTKEY.*

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