Audio precision Application note 3 in a series

LOUDSPEAKER TESTING WITH SYSTEM ONE

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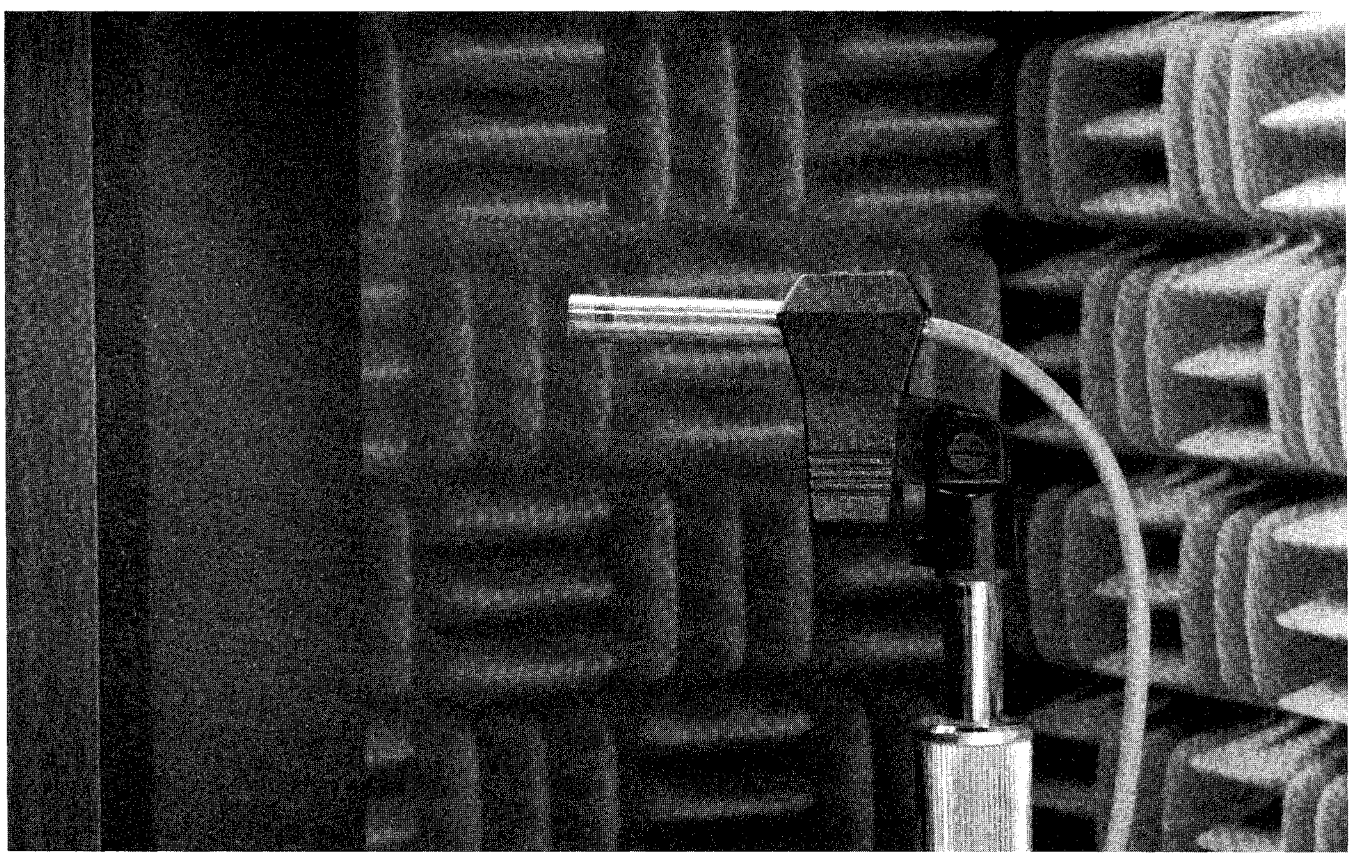


LOUDSPEAKER TESTING WITH SYSTEM ONE

Bob Metzler

Introduction

System One can accomplish the most commonly-used production tests for individual loudspeakers and complete loudspeaker systems. Many of the tests done during loudspeaker development can also be made with System One. Certain categories of development testing, such as swept second or third harmonic distortion across the frequency band and testing response semi-independently of acoustic reflections, will require System One's DSP module.

This loudspeaker testing note will briefly describe the following categories of tests with System One:

- Frequency response and sensitivity (efficiency), in relative or dBspl units
- Polarity
- Phase linearity and group delay
- Detection of undesired noises including rub & buzz, particles, cone breakup and cone cry, etc.
- Impedance testing, including Thiele-Small parameters.

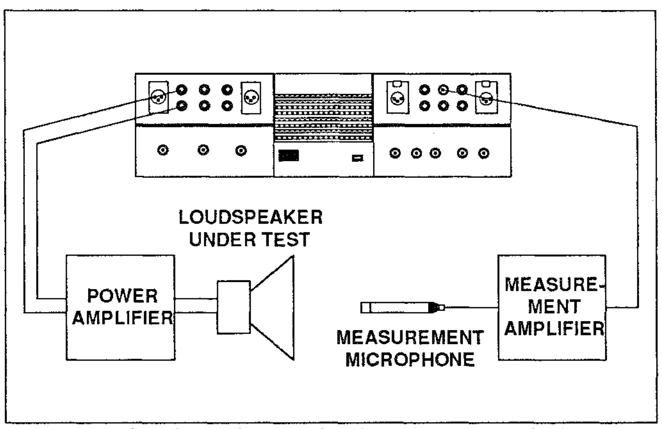


Figure 1 Hookup Diagram for Response, Sensitivity, Polarity, and Rub and Buzz Tests

Equipment Block Diagram

Response, rub & buzz, phase linearity, and polarity testing are all accomplished with the same basic equipment connections. These tests require several items in addition to System One; see Figure 1. The generator output is connected to the input of a power amplifier whose output drives the speaker under test. The power amplifier is necessary to provide the near-zero source impedance drive which most loudspeakers depend upon for normal operation.

A known-flat measurement microphone, typically one of the 1/2" or 1/4" Bruel & Kjaer models for full audio spectrum testing, is positioned in front of the speaker. These microphones are condenser types which require a polarizing voltage to operate. The dc voltage is typically provided by a power supply also furnished by B&K, which frequently is incorporated into a microphone preamplifier (measuring amplifier). The output of the microphone power supply or measuring amplifier is connected to the analyzer input of System One.

The sensitivity of System One for amplitude measurements is sufficient for use without the measurement amplifier at sound pressure levels typical for loudspeaker testing, but this signal amplitude may be insufficient for System One's phase meter. Phase linearity and the polarity testing method at low frequencies depend upon a phase measurement, thus the measurement amplifier may be necessary when these measurements are required.

Response Measurement Concepts

Frequency response measurement of loudspeakers is a much more complex topic than response measurements on amplifiers or most other electronic audio products. Near-field effects cause the response measured with close microphone spacing to vary from that with distant microphone placement. Loudspeakers are directional, so the response varies as the microphone is moved off the speaker axis. Unless the loudspeaker is being measured in an expensive anechoic chamber or in essentially free space outdoors, reflections of the sound waves will also arrive at the microphone and create standing wave patterns with cancellations and reinforcements. Multi-way loudspeaker systems, ported loudspeaker systems, and systems with some of the

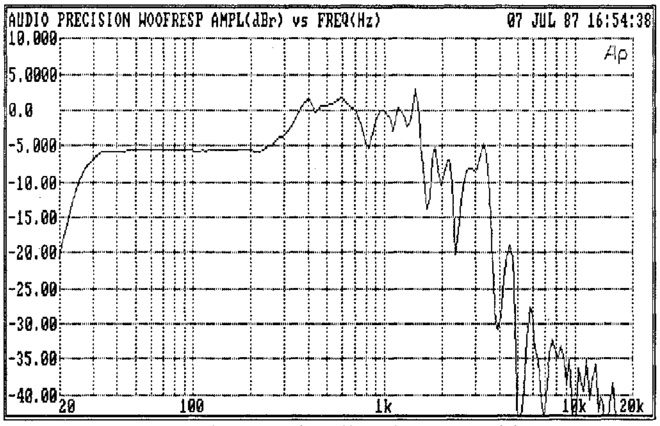


Figure 2 Anechoic Chamber Full-Spectrum Measurement of 15-Inch Woofer in Baffle

drivers facing the sides or rear of the cabinet all create a problem for measurement microphone placement—just where should the microphone be located so that it receives acoustical output which is a proper combination of the various drivers and ports? Furthermore, the end use of the speaker will not be in an anechoic chamber or laying on its back outdoors with listeners suspended above it—so what is the proper environment in which to measure it?

Known Good Standards (The "Golden" Unit)

There is no universally-accepted answer to the above questions for either development or production testing. Development normally includes a variety of measurements and measuring environments consistent with the philosophy of the designer, plus subjective listening tests. Production testing is normally simplified by an assumption: if a "known good speaker" can be selected by some combination of laboratory and listening tests, and if that known good speaker is measured in a practical, repeatable test environment, then other speakers of the same design which closely match those measurements ***in that same environment*** can be considered good units. Production testing of loudspeakers virtually never uses analytically-derived acceptance limits such as are used for amplifier testing. Instead, acceptance limits for testing of loudspeakers are experimentally derived from tests of one or several known good units. If the test chamber, test fixturing, or speaker-to-microphone placement is changed, the known good speaker must be remeasured and limits newly derived for the new setup.

Practical Response Tests

Figure 2 shows the result of a wide-range frequency response test on a large (15 inch diameter) woofer located in an anechoic chamber. A 200-step (201 point) sweep was used to obtain a large amount of detail. The rapid response variations above several hundred Hertz are presumably due to vibration modes across the large speaker cone.

Speed-Resolution Trade-Off

The number of steps selected for the sweep creates an almost-linear trade-off between data resolution and testing speed. Figures 3, 4, and 5 are response tests on a high-quality four-driver loudspeaker system in a very reflective space typical of most modern offices and laboratories. Figure 3 shows data for a 501 point sweep (approximately 1/50th octave steps), Figure 4 shows 201 points (1/20th octave), and Figure 5 71 points (about 1/8th octave steps). The additional resolution provided by the 500-step sweep is of little practical value in the production environment. The pronounced dip at about 2.2 kHz, for example, has a width of only 1% of center frequency and is clearly an artifact of room reflections.

A 70 step sweep is felt to be a reasonable compromise and will be used for examples through the remainder of this note. Note, however, that merely reducing the sweep (step) resolution does not guarantee that sharp nulls caused by reflections will be skipped across. A sweep of any resolution can coincidentally make one measurement at a response null caused by reflections. That point may then be interpreted as a "genuine" response dip when vectors are drawn from and to adjacent points.

Figure 6 shows the setup panels for SETTLED.TST, used to take the data in Figure 5. Connections are as shown in the block diagram of Figure 1, with the AUX input of the analyzer's channel A used as a convenience since the measurement amplifier's output cable is typically terminated in a type BNC connector. The power amplifier gain control is used to establish an appropriate acoustical level for the tests. Unbalanced output was selected for System One's generator since the power amplifier input was unbalanced. A vertical graphic scale of 50 dB was selected since the chart paper traditionally used for loudspeaker testing has a 50 dB range.

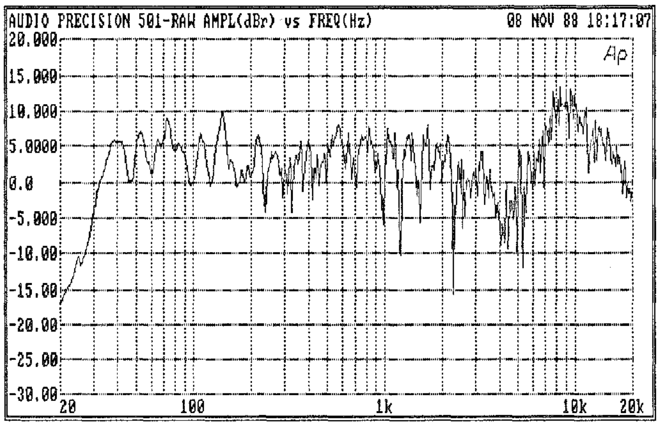


Figure 3 501-Point Measurement of Full-Range Speaker System in Reflective Space

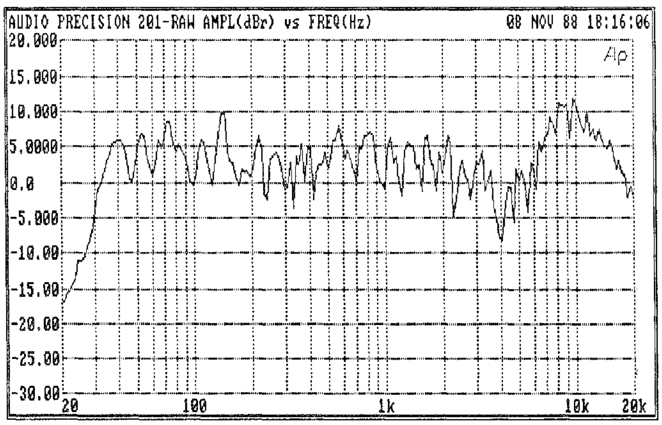


Figure 4 201-Point Measurement, Full-Range Speaker System in Reflective Space

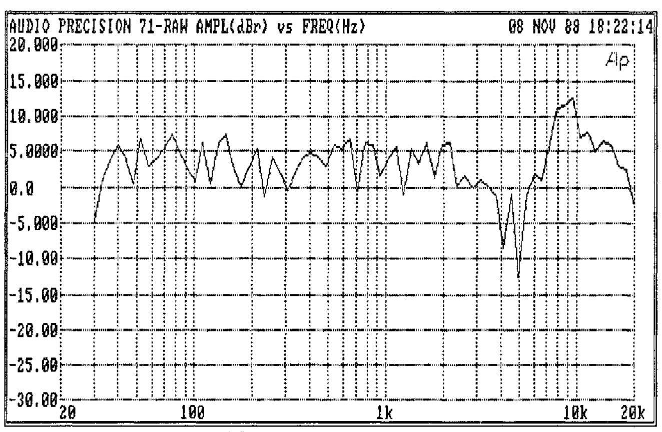
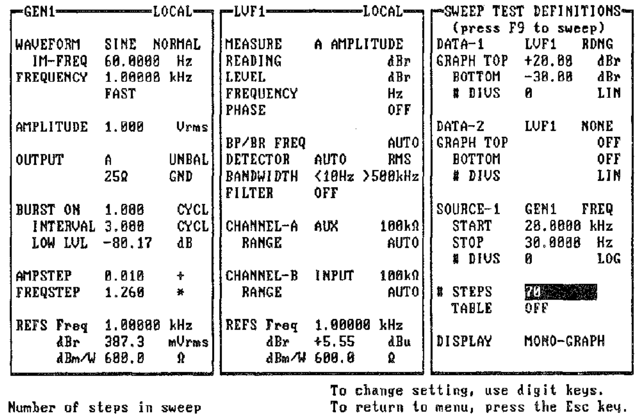


Figure 5 71-Point Measurement, Full-Range Speaker System in Reflective Space



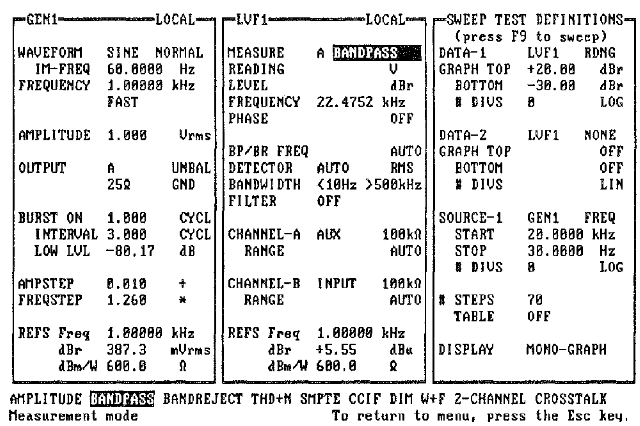
```
GEN1                 LOCAL   LVF1                 LOCAL   SWEEP TEST DEFINITIONS
                                                            (press F9 to sweep)
WAVEFORM   SINE   NORMAL    MEASURE    A AMPLITUDE        DATA-1      LVF1    RDNG
  IM-FREQ  60.0000  Hz      READING              dBr      GRAPH TOP   +20.00   dBr
FREQUENCY  1.00000 kHz      LEVEL                dBr        BOTTOM    -30.00   dBr
           FAST             FREQUENCY            Hz         # DIVS    0        LIN
                            PHASE                OFF
AMPLITUDE  1.000    Vrms                                  DATA-2      LVF1    NONE
                            BP/BR FREQ           AUTO     GRAPH TOP            OFF
OUTPUT     A        UNBAL   DETECTOR   AUTO      RMS        BOTTOM             OFF
           25Ω      GND     BANDWIDTH  <10Hz >500kHz        # DIVS             LIN
                            FILTER     OFF
BURST ON   1.000    CYCL                                  SOURCE-1    GEN1    FREQ
  INTERVAL 3.000    CYCL    CHANNEL-A  AUX       100kΩ      START     20.0000 kHz
  LOW LVL  -80.17   dB        RANGE              AUTO       STOP      30.0000  Hz
                                                            # DIVS    0        LOG
AMPSTEP    0.010    +       CHANNEL-B  INPUT     100kΩ
FREQSTEP   1.260    *         RANGE              AUTO     # STEPS     70
                                                            TABLE     OFF
REFS Freq  1.00000 kHz      REFS Freq  1.00000 kHz
     dBr   387.3   mVrms         dBr   +5.55     dBu      DISPLAY     MONO-GRAPH
     dBm/W 600.0   Ω             dBm/W 600.0     Ω

                                   To change setting, use digit keys.
Number of steps in sweep           To return to menu, press the Esc key.
```

Figure 6 Setup Panels, SETTLED.TST for Response Measurements

On the SWEEP SETTLING panel (not shown), the AMPLITUDE TOLERANCE was set to 3% (approximately 0.3 dB) which is adequate for the repeatability practical in loudspeaker testing. Settling is turned on (EXPONENTIAL) and input AUTOranging is selected.

Noise Susceptibility

Ambient noise levels are likely to be moderately high in practical production test environments. A degree of immunity to undesired noise can be obtained by using the BANDPASS mode rather than the AMPLITUDE mode of System One's READING meter during testing. BANDPASS mode inserts a 1/3 octave tunable filter in the measurement instrument. If AUTO is selected as the



```
GEN1                 LOCAL   LVF1                 LOCAL   SWEEP TEST DEFINITIONS
                                                            (press F9 to sweep)
WAVEFORM   SINE   NORMAL    MEASURE    A BANDPASS         DATA-1      LVF1    RDNG
  IM-FREQ  60.0000  Hz      READING              V        GRAPH TOP   +20.00   dBr
FREQUENCY  1.00000 kHz      LEVEL                dBr        BOTTOM    -30.00   dBr
           FAST             FREQUENCY  22.4752 kHz          # DIVS    0        LOG
                            PHASE                OFF
AMPLITUDE  1.000    Vrms                                  DATA-2      LVF1    NONE
                            BP/BR FREQ           AUTO     GRAPH TOP            OFF
OUTPUT     A        UNBAL   DETECTOR   AUTO      RMS        BOTTOM             OFF
           25Ω      GND     BANDWIDTH  <10Hz >500kHz        # DIVS             LIN
                            FILTER     OFF
BURST ON   1.000    CYCL                                  SOURCE-1    GEN1    FREQ
  INTERVAL 3.000    CYCL    CHANNEL-A  AUX       100kΩ      START     20.0000 kHz
  LOW LVL  -80.17   dB        RANGE              AUTO       STOP      30.0000  Hz
                                                            # DIVS    0        LOG
AMPSTEP    0.010    +       CHANNEL-B  INPUT     100kΩ
FREQSTEP   1.260    *         RANGE              AUTO     # STEPS     70
                                                            TABLE     OFF
REFS Freq  1.00000 kHz      REFS Freq  1.00000 kHz
     dBr   387.3   mVrms         dBr   +5.55     dBu      DISPLAY     MONO-GRAPH
     dBm/W 600.0   Ω             dBm/W 600.0     Ω

AMPLITUDE BANDPASS BANDREJECT THD+N SMPTE CCIF DIM W+F 2-CHANNEL CROSSTALK
Measurement mode                   To return to menu, press the Esc key.
```

Figure 7 Setup Panels, BANDPASS.TST for Better Ambient Noise Rejection During Response Tests

BP/BR control mode, the filter center frequency will automatically track the generator frequency during a sweep. The effect of any spectral components of noise away from the present generator frequency will then be attenuated. Figure 7 shows the setup panels for BANDPASS.TST.

BANDPASS mode has its costs, however. The Q of the bandpass filter will cause ringing at each frequency step, so the data will take longer to settle to within the specified tolerance. A frequency sweep in BANDPASS mode will thus take somewhat longer than in AMPLITUDE mode. The increase in time will depend upon the actual step-to-step response variations of the loudspeaker and room.

Standing Waves, Reflections, and Smoothing

Neither an anechoic chamber nor outdoor free-space testing are practical for production testing. In most test locations, acoustical reflections and ambient acoustical noise are facts of life. Particularly at the higher frequencies, the acoustical energy arriving at the microphone is a combination of the direct signal from speaker to microphone plus a large number of reflections from walls, ceiling, floor, human operators, etc. In a typical office or laboratory environment when using high-frequency sinewave signals (10 kHz, for example), movement of a human being several feet away from the microphone can cause several dB change in the measured value. Changes on the order of a few inches in microphone or loudspeaker placement can cause response curve variations of many dB at high frequencies. Even at fixed speaker and microphone locations and with no people moving within a reasonable distance, a response curve may show individual sharp peaks and dips of 10 to 20 dB due to standing wave effects.

For reasonable results, some degree of acoustical treatment of the measuring location should be done. Commercially-available sound absorption materials should be applied to wall, ceiling, and floor surfaces. It may be effective to design the shape of the test chamber as something other than a rectangle to avoid high-intensity single-bounce reflections from the back wall into the measurement microphone.

Since it is not economically practical to provide a true anechoic chamber for most testing, speaker measurements in a reflective space are almost invariably smoothed by one technique or another to improve measurement repeatability and reduce the visible effect of reflections to a value more consistent with a human's ability to hear them. Even simple analog measurement techniques, where no "smoother" appears as such on a block diagram, do in fact typically smooth due to the sweep rates and measurement response times used. Measurement instruments have some finite time constant or response time; they cannot instantaneously follow a rapidly-changing signal amplitude. If the measurement instrument has frequency selectivity (spectrum analyzer or tracking filter), the response time is even slower due to the energy storage or ringing effect of the tuned filter. When the stimulus signal is swept rapidly, the measurement instrument simply cannot follow the more rapid, extreme variations and the displayed result is smoothed—peaks are reduced and dips are filled in.

System One software, when settling is enabled, pauses long enough at each frequency step for the measured value to stabilize to the tolerance specified by the user (about 0.3 dB in the setup of Figure 6). The graph thus obtained will show the extreme peaks and dips due both to actual loudspeaker response variations and room reflections. Figure 5 shows the data immediately at the end of the test, before smoothing.

Computational Smoothing

System One's COMPUTE SMOOTH feature can be invoked following a test to provide any desired amount of data smoothing. The graph shown in Figure 8 is the result of a single pass of smoothing. That is:

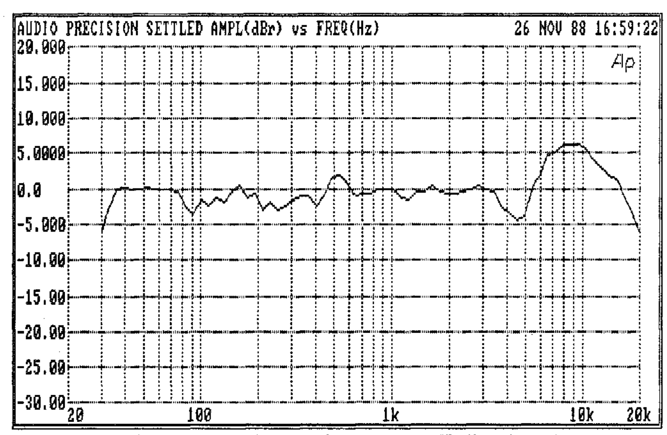


Figure 8 *Data From Test of Figure 5, Following One Iteration of Smoothing*

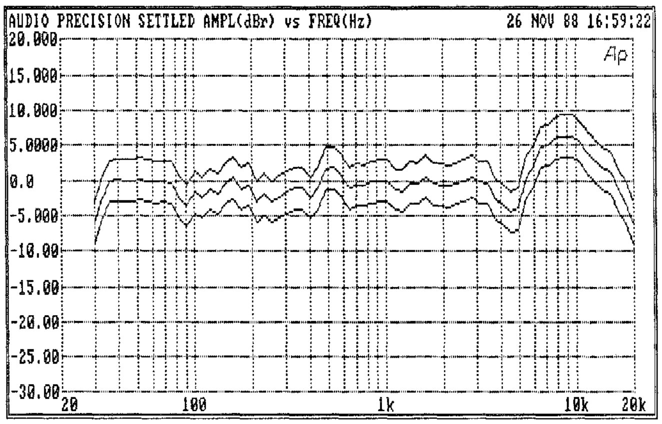


Figure 9 Data With Smoothed Upper and Lower Acceptance Limits, Offset ± 3 dB

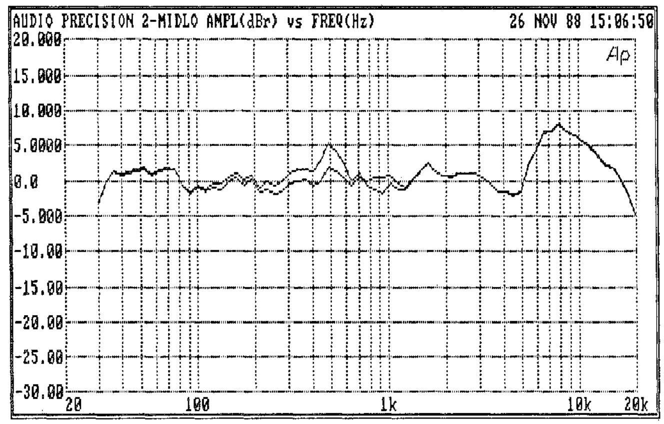


Figure 10 Two Response Tests Superimposed, One With Defective Lower-Midrange Driver

COMPUTE SMOOTH 1,1 <Enter>

was invoked following the test to cause the system to select DATA-1 and make one smoothing pass through the data. More iterations of smoothing will be required on higher-resolution sweeps to result in comparably smoothed results.

With 70 steps (71 points), one pass (COMPUTE SMOOTH 1,1) yields curves with detail similar to curves obtained by purely analog sweep test systems at the speeds typically used in loudspeaker production tests. Using one pass smoothing to generate acceptance limits and one pass smoothing on the data before limits comparison provides a degree of protection from very narrow-band peaks and dips. This permits speaker-to-microphone movement of several inches in typical testing environments before the response varies by more than the 2 or 3 dB typically permitted. If the test location is reasonably well designed with sound absorption material, careful microphone placement, and a test "chamber" design that minimizes reflections to the microphone locations, smoothing may not be necessary.

Figure 10 shows two response tests superimposed, both after smoothing. In one case, known good speakers were installed for all four drivers. In the second test, a lower-midrange driver (covering the range between approximately 400 Hz and 800 Hz) was installed which had a peaked frequency response. The peak at 500 Hz fails the upper limit shown in Figure 9 and would cause this speaker to be rejected.

Acceptance Test Limits

Figure 9 shows the same graph as Figure 8, plus upper and lower test limits. The limits were generated by making a curve on the known good speaker, doing one pass of smoothing, and using COMPUTE NORMALIZE to offset the curve for use as a limit. The curve was offset to a value of +3 dBr and saved as UP.LIM. COMPUTE NORMALIZE was used again to offset it down to -3 dBr and the result saved as LO.LIM. The test was re-loaded and the NAMES UPPER and NAMES LOWER commands used to "attach" the limit files to the test.

Forming Limits from the Average of Several Tests

Acceptable reference speakers are typically selected by listening tests from a pilot production quantity. All good units are not identical to one another. It is frequently desirable to derive limits from the average of a number of acceptable units, rather than offsetting measurements on a single speaker for use as upper and lower limits.

Audio Precision has created a BASIC program (APAVG.BAS) which takes data from a number of measurements and averages them. The program asks for the number of dB which upper and lower limits should be offset from the average. It saves three files; an upper limit (UPPER.DAT), lower limit (LOWER.DAT), and the average data (AVGFILE.DAT). These files can be imported to System One software via the LOAD DATA command and saved as .LIM files. This program can be found on the companion diskette to this applications note.

Several conditions must be true for this program to work properly. All the .DAT files to be averaged must consist of measurements at identical frequencies. Thus, they must have been saved as a result of running the identical .TST file on several loudspeakers, with no changes in START frequency, STOP frequency, number of steps, direction of sweep, or LOG/LIN choice. The vertical display unit must be a dB unit (dBu, dBV, or dBr). The program will prompt for the name of a master .TXT file which describes the number of steps and lists file names to be averaged; this file can be created using the Edit Comments capability of S1.EXE. The program prompts for a value to offset the average upwards to create UPPER.DAT and another value to offset downwards to create LOWER.DAT.

Limits Comparison After Smoothing

If smoothing is to be used, data must not be compared to limits until after smoothing. This can be accomplished in a procedure in at least two ways. The test can be stored with no limit files attached. Following the test (F9) and COMPUTE SMOOTH operation, the procedure can name upper and lower limits and then re-graph data (F7), which causes a limits comparison of the smoothed data. This section of a procedure would thus appear:

```
LOAD TEST XXXXX/R
/F9/E
COMPUTE SMOOTH 1,1/R
NAMES UPPER UPLIMIT/R
NAMES LOWER LOWLIMIT/R
/F7/E
IF ERROR[ action to be taken on failure]
```

Or, the test can be stored with limits attached but no error file named. The test can be loaded and run, COMPUTE SMOOTH used, NAMES ERROR-FILE used to specify an error file for pass/fail data to be written into, and a re-graph done to compare smoothed data to limits:

```
LOAD TEST XX/R
/F9/E
COMPUTE SMOOTH 1,1/R
NAMES ERROR-FILE FILENAME/R
/F7/E
```

Test Speed

In high-volume production testing, the speed of a test can be very important. Operating with autoranging and settling turned on is not the fastest mode for System One to test loudspeakers. However, simply turning settling off will not yield acceptable data quality. Transients occur at each generator step. Some of the transients are the result of amplitude range switching in the front end of the analyzer. Selecting an appropriate fixed range eliminates this source of transients. The fixed range must be chosen so as not to cause clipping on the signal from the microphone at the highest-output portion of the loudspeaker spectrum. This can be determined by first using SETTLED.TST, noting the frequency of maximum response (***before smoothing***), setting the generator to that frequency, and then changing the analyzer from AUTO to fixed range. It is even safer to further select the next higher range. System One's amplifiers and detectors following the input section have over 40 dB usable linear dynamic range, so it is better to be conservative in range selection and absolutely guarantee that clipping will not occur in the analyzer. Figures 11 and 12 show setup panels and the data for NO-SETTL.TST.

Transients will still occur in the generator and analyzer, particularly when the signal passes through the frequency range-switching points just above 2 kHz and 200 Hz. Choosing a sufficiently-long DELAY on the settling panel will inhibit measurements until the transients have died away. This DELAY time must be selected experimentally. The recommended process is as follows:

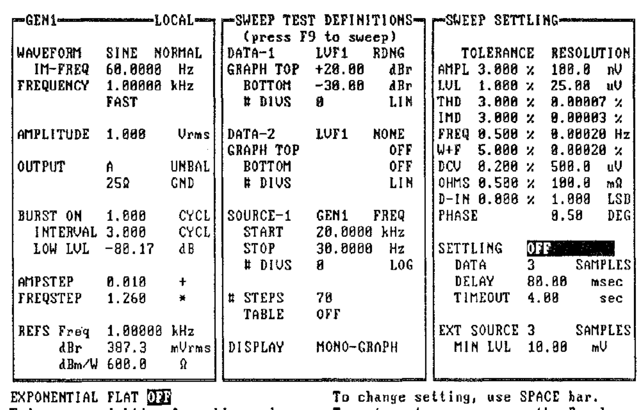


Figure 11 *Setup Panels, NO-SETTL.TST for Faster Testing. Analyzer Panel (not shown) Same as SETTLED.TST*

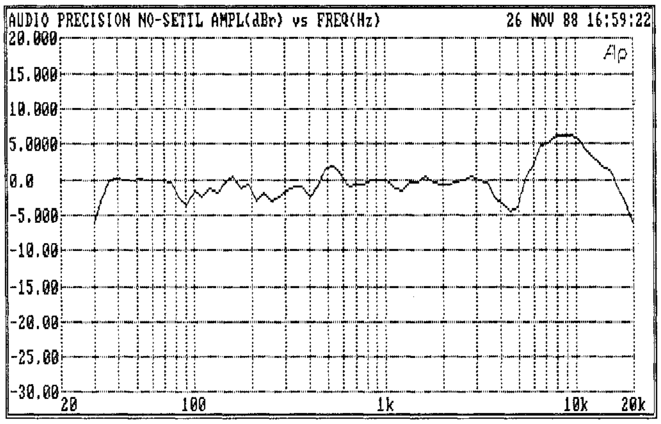


Figure 12 Response Graph (After Smoothing), NO-SETTL.TST for Faster Testing

1. Make a frequency response curve with SETTLED.TST, which uses settling.

2. At the end of the test, press <Alt>F8 to store a video image of that graph into memory.

3. Select the fixed range as described above.

4. Select a starting value of DELAY such as 50 ms.

5. Press F8 to restore the stored image, followed by F9 to make a new test (without settling) onto the same screen.

6. If the two graphs do not repeat at all points within the desired repeatability, increase the DELAY value and repeat step 5. Again, it is best to be conservative and use a value of DELAY slightly longer than the minimum determined by experiment. Values around 80 milliseconds are typically sufficient with the READING meter in AMPLITUDE mode. Slightly longer delays may be necessary in BANDPASS mode, as in BANDPASS.TST.

NO-SETTL.TST will typically run in about two-thirds the time of SETTLED.TST. With 71 points, for example, SETTLED.TST typically runs in about 18 seconds while NO-SETTL.TST runs in about 12 seconds.

Testing speed can also be increased by using the DISPLAY NONE setting at the bottom of the SWEEP TEST DEFINITIONS panel rather than graphic display.

Absolute Sound Pressure Measurements (dBspl)

It is sometime necessary to measure the sound pressure level of loudspeakers and other acoustic transducers in absolute terms. The standardized unit is dBspl, or dB sound pressure level. Precision measurement microphones are normally supplied by their manufacturer with calibration data including a sensitivity specification for the specific serial number microphone. A typical specification, quoted from a calibration chart for a Bruel & Kjaer model 4136 1/4" microphone, reads:

-80.3 dB re. 1 V/μbar

The specification further states that this measurement was made at 250 Hz, 760 mm atmospheric pressure, with a 200 Volt polarizing voltage, and was measured at the cathode-follower output of the microphone into a load of 50 kilohms or higher. The cathode follower (or transistorized impedance matching circuit, in some microphones) is inside the microphone assembly. The specification thus means that with the microphone output connected into a high impedance load such as System One's analyzer input or a B&K measuring amplifier input, the microphone output would be -80.3 dBV if the microphone were immersed in a one μbar sound pressure field.

To use System One's dBr unit as dBspl, the 0 dBspl value for the specific microphone must be computed and entered into the REF dBr field near the bottom of the LVF1 panel. Zero dBspl is defined as 0.0002 μbar, which is 74 dB below one μbar. In this case, adding the -80.3 dBV microphone sensitivity and the -74 dB difference between zero dBspl and one μbar gives a -154.3 dBV result. Enter -154.3 dBV into the REF dBr field, select dBr for the LVF1 panel display and/or DATA-1 units for a sweep test, and the values will be displayed directly in dBspl.

If a measurement amplifier is used between the microphone output and System One input, the REF dBr field must be further corrected by the gain of the measurement amplifier. If, for example, a measurement amplifier is set for 60 dB gain and used with the microphone of this example, the REF dBr value must be changed from -154.3 dBV to -94.3 dBV.

Sensitivity (Efficiency) Tests

Sensitivity is commonly evaluated as part of the frequency response test. The upper and lower limits applied to a response test are normally absolute and thus would reject a speaker whose response curve *shape* was acceptable, but which fell below the lower limit due to low sensitivity. If it is desired to make a discrete test for sensitivity during a procedure, it may be done with either sine wave or pink noise stimulus.

Sine wave sensitivity testing is frequently done at lower-midrange frequencies in the 300-500 Hz region. These frequencies are in the general spectral area most closely relating to human perceptions of loudness. They are also less sensitive to room reflection problems than are higher frequencies. However, if any significant reflections exist, a single-point measurement is susceptible to error which could cause rejection of good units.

A better solution is use of a signal such as pink-noise or 1/3 octave band-limited noise. The non-coherent nature of such signals essentially eliminates the standing wave problem. System One's BUR option includes the ability to generate random or pseudo-random, pink or white noise, or 1/3 octave bandwidth limited pink noise in the BPASS (bandpass) mode. The FREQUENCY field of the GEN1 panel selects the center frequency of the bandpass filter in the noise generator. Figure 13 shows the setup panels for SENSITIV.TST which uses bandpassed pink noise centered at 315 Hz as the signal.

```
GEN1                  LOCAL   LVF1                  LOCAL   SWEEP TEST DEFINITIONS
                                                              (press F9 to sweep)
WAVEFORM   PSEUDO BPASS       MEASURE    A AMPLITUDE        DATA-1     LVF1   RDNG
  IM-FREQ  60.0000   Hz       READING              dBr      GRAPH TOP  +20.00  dBr
FREQUENCY  315.000   Hz       LEVEL                dBr        BOTTOM   -30.00  dBr
           FAST               FREQUENCY            Hz         # DIVS   0       LIN
                              PHASE                OFF
AMPLITUDE  1.000     Vrms                                   DATA-2     LVF1   NONE
                              BP/BR FREQ           AUTO     GRAPH TOP          OFF
OUTPUT     A         UNBAL    DETECTOR   AUTO      RMS        BOTTOM           OFF
           25Ω       GND      BANDWIDTH  <10Hz >500kHz        # DIVS           LIN
                              FILTER     OFF
BURST ON   1.000     CYCL                                   SOURCE-1   GEN1   NONE
  INTERVAL 3.000     CYCL     CHANNEL-A  AUX       100kΩ      START            OFF
  LOW LVL  -80.17    dB         RANGE              AUTO       STOP             OFF
                                                              # DIVS   0       LOG
AMPSTEP    0.010     +        CHANNEL-B  INPUT     100kΩ
FREQSTEP   1.260     *          RANGE              AUTO     # STEPS    0
                                                              TABLE    OFF
REFS Freq  1.00000 kHz        REFS Freq  1.00000 kHz
     dBr   387.3   mVrms           dBr   -4.13     dBu      DISPLAY    MONO-GRAPH
     dBm/W 600.0   Ω               dBm/W 600.0     Ω

 WHITE  PINK  BPASS  EQBPN                  To change setting, use SPACE bar.
Waveform modifiers                          To return to menu, press the Esc key.
```

Figure 13 *Setup Panels, Sensitivity Test Using 1/3 Octave Pink Noise*

Polarity Testing

A common error in manufacturing of loudspeakers and loudspeaker systems is to connect voice coils to the terminals with reversed polarity. Any individual driver in a multi-way speaker system can also accidentally be wired with incorrect polarity. Even when all drivers are correctly phased with respect to one another, it is possible to have the interior wiring to the external connection terminals of the cabinet reversed. An individual driver reversed will cause a dip in frequency response near the crossover frequency to the adjacent driver, since the two speakers are then producing acoustical output of nearly identical amplitudes but out of phase. An entire system wired out-of-phase would presumably be indetectable in a monaural application, but unacceptable in stereo systems.

Two different methods of determining correct polarity are practical with System One. At high frequencies, the frequency response measurement itself will detect individual mis-wired drivers. The cancellation caused by out-of-phase operation of the two drivers in the crossover frequency region is easily measureable. Figure 14 shows superimposed response graphs made of a speaker system with the tweeter in correct and reversed polarity. The sharp dip at 5 kHz on the reverse polarity graph will easily cause rejection by the frequency response limits. The nominal value of the crossover frequency of this particular system is just above 3 kHz. The dip will not necessarily appear exactly at the crossover frequency, both because the measurement microphone may not be exactly equidistant from the two drivers and because acoustic reflections are also part of the measurement. Care should be taken to

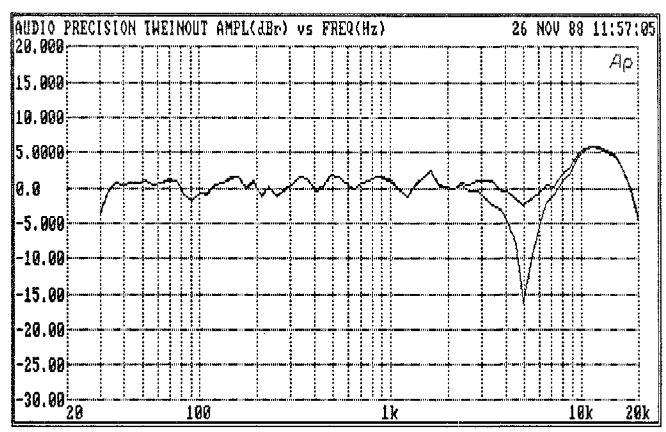


Figure 14 *Superimposed Graphs, Tweeter in Correct and Incorrect Polarity*

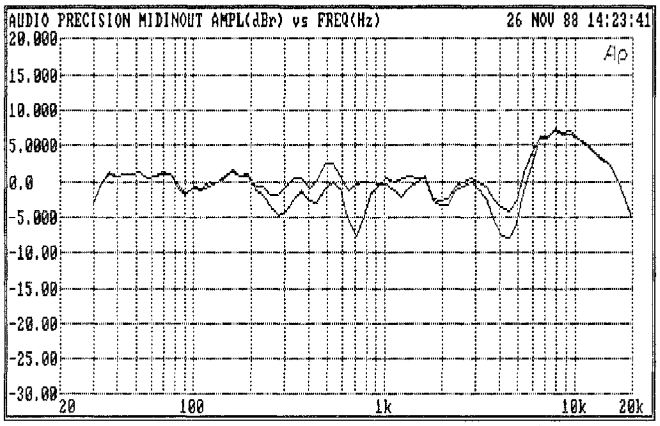


Figure 15 Superimposed Graphs, Mid-range Driver in Correct and Reversed Polarity

locate the measurement microphone as nearly equidistant as possible from the two highest-frequency drivers. A one-inch difference in path length from the two drivers to the mic would result in cancellation at a 6.6 kHz frequency if the drivers were in-phase.

Figure 15 shows superimposed graphs of the speaker system, with the mid-range driver connections reversed in one case. The dips at 700 Hz and 4.5 kHz would both be rejected by the lower response limits. The nominal crossover values of this particular system are 800 Hz and 3 kHz.

Testing for proper polarity of woofers and for polarity inversion of the entire system is best done by a different technique. Figure 16 shows the measurement result (single-point measurement, hence tabular display) and Figure 17 shows the setup panels. By selecting GEN-MON for analyzer channel B and turning on both the A and B output channels of the generator, the generator output is automatically routed to the analyzer B input to serve as a reference for the phase meter. The generator frequency is set to a value near the center of the woofer response range (100 Hz in the example shown). Selecting the A channel connects the READING meter and one phase meter input to the output of the measurement microphone. Signal level must be at least 10 millivolts for phase measurements, which usually requires a microphone preamplifier or measurement amplifier. The phase meter makes a phase measurement of the complete generator-to-microphone path. The measurement is not an absolute measurement, but must be compared to a known good system. A polarity reversal will cause an approximate 180 degree change in the measurement, so acceptance limits on the order of ± 30 degrees will easily reject mis-polarized units

```
WOOFPOLR   26 NOV 88   15:20:29
NONE(OFF)        PHASE(deg3)
     OFF    +56.1      deg3
```

Figure 16 Tabular Data Display, Woofer Polarity Test

```
GEN1                    LOCAL   LVF1                     LOCAL   SWEEP TEST DEFINITIONS
                                                                   (press F9 to sweep)
WAVEFORM   SINE     NORMAL      MEASURE    A AMPLITUDE           DATA-1     LVF1    PHASE
  IM-FREQ  60.0000  Hz          READING                dBr       GRAPH TOP  +360.0  deg3
FREQUENCY  100.000  Hz          LEVEL                  dBr         BOTTOM   +0.0    deg3
           FAST                 FREQUENCY              Hz          # DIVS   0       LIN
                                PHASE                  deg3
AMPLITUDE  1.000    Vrms                                         DATA-2     LVF1    NONE
                                BP/BR FREQ             AUTO      GRAPH TOP          OFF
OUTPUT     A&B      UNBAL       DETECTOR   AUTO        AVG         BOTTOM           OFF
           25Ω      GND         BANDWIDTH  <10Hz >500kHz           # DIVS           LIN
                                FILTER     OFF
BURST ON   1.000    CYCL                                         SOURCE-1   GEN1    NONE
  INTERVAL 3.000    CYCL        CHANNEL-A  AUX         100kΩ       START            OFF
  LOW LVL  -80.17   dB            RANGE                AUTO        STOP             OFF
                                                                   # DIVS   0       LOG
AMPSTEP    0.010    +           CHANNEL-B  GEN-MON     100kΩ
FREQSTEP   1.260    *             RANGE                AUTO      # STEPS    0
                                                                   TABLE    OFF
REFS Freq  1.00000  kHz         REFS Freq  1.00000 kHz
      dBr  387.3    mVrms             dBr  1.308   V             DISPLAY    MONO-GRAPH
    dBm/W  600.0    Ω               dBm/W  600.0   Ω

OFF A  B  A&B  A&-B                        To change setting, use SPACE bar.
Output OFF/ON                              To return to menu, press the Esc key.
```

Figure 17 Setup Panels, Woofer Polarity Test. Note Both Generator Channels ON, GEN-MON Connection on Analyzer

while allowing for normal unit-to-unit variations and small variations in speaker-to-microphone distance. This test is stored as POLARITY.TST on the diskette.

Note that this measurement is a measurement of total phase delay, including the delay due to propogation through air from the speaker to the microphone. This technique is thus useful only at lower frequencies, where the unit-to-unit change in speaker-to-microphone distance as successive speakers are positioned by the operator results in phase changes within practical acceptance limits. For example, at 1 kHz the acoustical wavelength is approximately 13 inches. Since one wavelength corresponds to a phase rotation of 360 degrees, one inch is 28 degrees. To pass a ± 30 degree tolerance at 1 kHz would thus require holding the spacing tolerance to one inch.

The recommended polarity testing process consists of detecting mis-wired high frequency drivers, perhaps above 500 Hz, by the resulting dips in frequency response. Mis-wired woofers and a reverse-wired complete cabinet will be found by the input-output phase measurement technique at low frequencies.

Phase Linearity and Group Delay

Recent years have seen a growth in interest in phase linearity (or constant group delay) of all portions of audio systems, and loudspeakers are no exception. Although easy to perform, these tests are unlikely to be done on a production line basis. They are becoming more common during the engineering development process.

Phase Linearity

Phase linearity is a simple extension of the phase delay measurement process described above for detecting polarity errors at low frequencies. Instead of a single "spot" measurement of generator-to-microphone phase

```
-GEN1------------LOCAL-  -LVF1--------------LOCAL-  -SWEEP TEST DEFINITIONS-
                                                      (press F9 to sweep)
WAVEFORM   SINE  NORMAL  MEASURE    A AMPLITUDE     DATA-1      LVF1   PHASE
  IM-FREQ  60.0000  Hz   READING              dBr   GRAPH TOP   +0.0     deg
FREQUENCY  .900000 kHz   LEVEL                dBr     BOTTOM    -4000.0  deg
           FAST          FREQUENCY            Hz      # DIVS    0        LIN
                         PHASE                deg
AMPLITUDE  1.000   Vrms                             DATA-2      LVF1   NONE
                         BP/BR FREQ           AUTO  GRAPH TOP            OFF
OUTPUT     A&B    UNBAL  DETECTOR   AUTO      AVG     BOTTOM             OFF
           25Ω    GND    BANDWIDTH  <10Hz >500kHz     # DIVS             LIN
                         FILTER     OFF
BURST ON   1.000   CYCL                             SOURCE-1    GEN1   FREQ
  INTERVAL 3.000   CYCL  CHANNEL-A  AUX     100kΩ     START     100.000  Hz
  LOW LVL  -80.17  dB      RANGE              AUTO    STOP      5.00000 kHz
                                                      # DIVS    0        LIN
AMPSTEP    0.010   +     CHANNEL-B  GEN-MON 100kΩ
FREQSTEP   1.260   *       RANGE              AUTO  # STEPS     50
                                                      TABLE     OFF
REFS Freq  1.00000 kHz   REFS Freq  1.00000 kHz
     dBr   387.3  mVrms       dBr   1.308   V       DISPLAY     MONO-GRAPH
     dBm/W 600.0   Ω          dBm/W 600.0   Ω

deg deg1 deg3 rad                       To change setting, use SPACE bar.
Primary units                           To return to menu, press the Esc key.
```

Figure 18 Setup Panels, Input-Output Phase Test

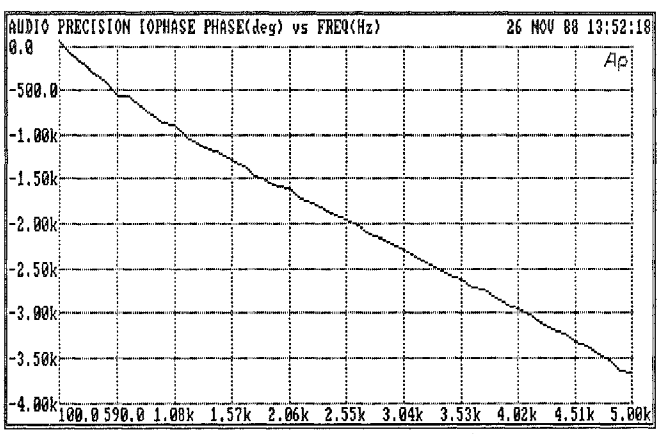


Figure 19 Input-Output Phase, Loudspeaker to Measurement Microphone

delay, the generator is stepped in frequency across the spectrum of interest. Figure 18 shows the setup panels for such a test across the 100 Hz-5 kHz spectrum, stored as IOPHASE.TST on the diskette. A linear sweep was selected so that a linear phase response (constant group delay) will plot as a straight line. Figure 19 is the resulting graph of phase versus frequency on the four-way loudspeaker system used for most of the examples in this note.

Note that System One has three different degree unit selections. "Deg1" and "deg3" select fixed ±180 or 0 to +360 ranges, respectively. The "deg" unit used in this test automatically keeps track of complete 360 degree phase rotations to produce a continuous graph rather than one with abrupt graphic "re-sets" every 360 degrees. For this feature to work properly, the phase change per frequency step cannot exceed 180 degrees. This requires that the computed period of the frequency step size must be at least twice the acoustical delay. For example, with a two millisecond acoustical delay (approximately two foot spacing), a computed minimum period of four milliseconds is required and the step size in the frequency sweep consequently must be less than 250 Hz.

While Figure 19 shows that the this speaker system to have an approximately linear phase response, it is difficult to see small variations when the vertical axis extends across 4,000 degrees. System One's COMPUTE LINEARITY menu function can be used to improve the analysis. COMPUTE LINEARITY will fit the best straight line to the measured data or a specified portion of the data, and then subtract each data point from that best straight line. The resulting graph (Figure 20) is a display

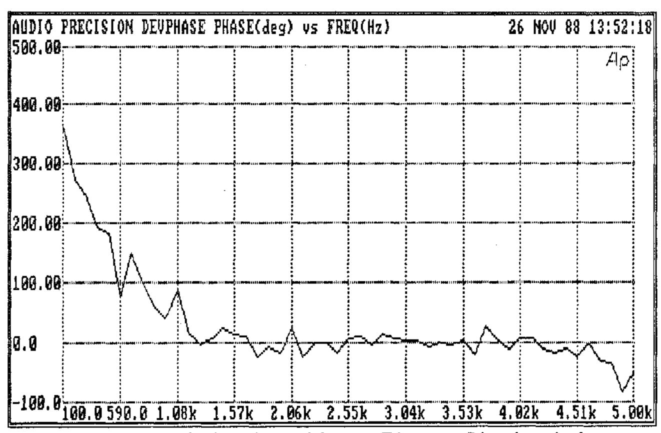


Figure 20 Deviation from Linear Phase, Obtained via COMPUTE LINEARITY Function

of deviation from perfect linearity, or from linear phase in this case. COMPUTE LINEARITY was invoked with the 2 kHz to 4 kHz portion of the data specified as the "reference" area, simply because the linearity looked best in that region. Following this operation, the DATA-1 GRAPH TOP and BOTTOM were changed to better display the deviation from perfect phase linearity.

Group Delay

Another approach is to use the phase versus frequency data to compute group delay. Bruce Hofer has created a procedure called GRPDEL.PRO which incorporates a compiled BASIC program called GRPDEL.EXE. These files are included on the diskette available with this application note. Figure 21 shows the result of running this procedure on the data of Figure 19. The group delay technique has the additional advantage of giving the correct absolute delay if the frequency step size—acoustical delay relationship complies with the requirements described above under Phase Linearity. Linear sweep should be selected to take the data. The graph for display of data may be linear or log, as desired. In the example shown, the distance from the center of the speaker system to the microphone was approximately two feet, yielding an expected delay of about two milliseconds. The graph shows this value of group delay at higher frequencies, rising to a slightly higher value at the lowest frequencies. A portion of this increase may be due to the longer acoustical path, since the woofer was near the bottom of the "tower" style cabinet.

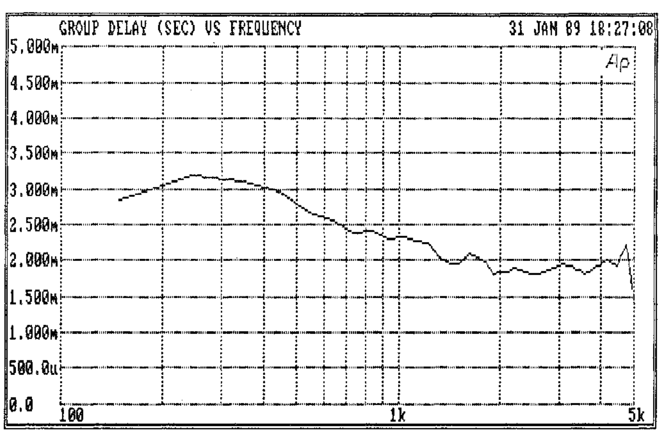


Figure 21 *Group Delay, Multi-Way Loudspeaker System*

Rub and Buzz Tests

Testing for rub & buzz, particles, cone cry, and other such noises is probably the most difficult part of production testing of loudspeakers. Many of these noises are only generated when the loudspeaker is being driven at quite high levels, since they often are caused by the voice coil hitting foreign objects or the frame at peak excursions. Depending on the particular noise-generating mechanism, the resulting noise may consist of high order harmonics of the stimulus frequency or of a broader frequency band of noise.

In some cases, these defects can be caught by a straightforward THD+N measurement as the generator frequency is swept. Many times, however, quite offensive audible noise is the result of very high order harmonics which are lower in amplitude than the normal (and relatively harmless) 2nd and 3rd harmonics. In such a case, the THD+N measurement is dominated by the 2nd and 3rd harmonics present from even a good loudspeaker, with the energy from an ugly-sounding high order harmonic insufficient to produce much change in the THD+N reading.

The most reliable technique for detecting rub & buzz and other similar noises with System One involves use of the BANDPASS mode. The generator is typically swept across a range of 2-3 octaves or a decade while the measurement microphone output is measured through System One's BANDPASS filter which is fixed in frequency. The bandpass filter may be fixed at approximately three to six times the upper frequency end of the sweep. Higher positioning results in less sensitivity to second and third harmonic distortion of normally-functioning loudspeakers. Each different loudspeaker design is likely to have its rub and buzz energy concentrated at different parts of the spectrum, requiring some degree of experimentation with known good and bad speakers to choose an optimum frequency for the bandpass. Typical response of the bandpass filter is -42 dB at 1/3 center frequency and -32 dB at 1/2 center frequency.

Figure 22 shows the setup panels for such a test on the tweeter of the multi-way speaker system. This test is stored as RUB&BUZZ.TST on the diskette. Most rub & buzz noises are most likely to be stimulated by driving the speaker through the lower portion of its operating range (even below resonance in the case of woofers). For this tweeter, whose operating range is from the 3 kHz crossover to the upper limits of audibility, the sweep was set

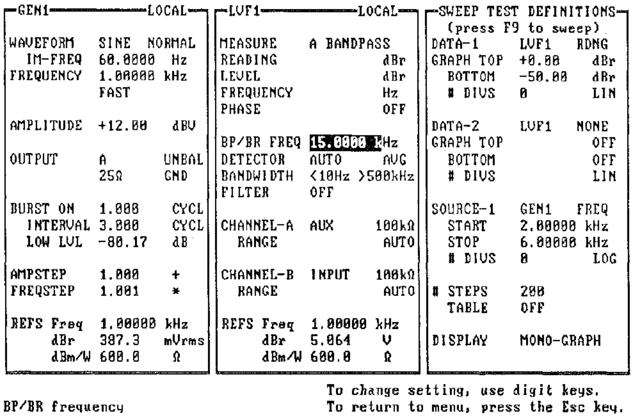


Figure 22 *Setup Panels, Rub and Buzz Test for Tweeter*

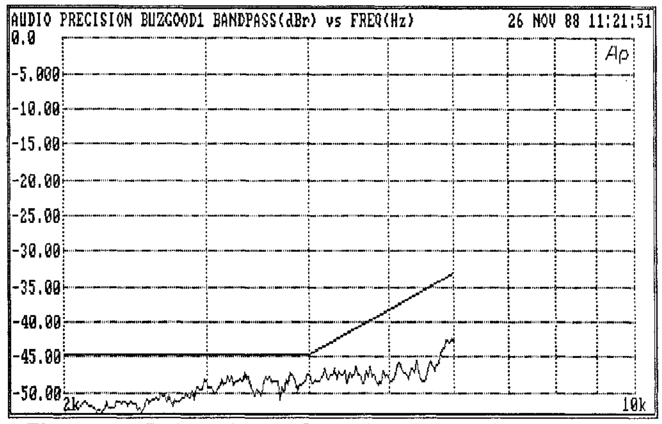


Figure 23 *Rub and Buzz Test Result, Good Tweeter #1*

from 2 kHz to 6 kHz. The generator amplitude was set 12 dB higher than the level used for response testing; the increase in amplitude for any particular driver will depend on its design, specifications, and concern for the operators working in the vicinity. The BP/BR filter frequency was fixed at 15 kHz, some 7 times higher than the sweep start frequency and 2.5 times the sweep stop. For test development and illustration purposes, a very high-resolution sweep (200 steps) was used. In actual production testing, resolution on the order of 1/3 to 1/10th octave (10 to 30 steps for the sweep range shown) seems to be sufficient to catch the most narrow-band noise effects commonly observed.

Figure 23 shows the resulting measurement curve on a good (no buzz) speaker. Note that the curve is ***not*** the frequency response of the speaker. Instead, it represents the energy falling into a passband centered at 15 kHz (-3 dB bandwidth of about 20% of center frequency) as the speaker is driven with a sweep through a much lower band of frequencies. The 0 dB reference is selected as the average response of the speaker near 15 kHz (BP center frequency). This reference setting needs to be done only once for each different speaker design to be tested--not for every unit. The gradual increase in amplitude at the right of the curve is a combination of normal harmonic energy falling into the filter passband, plus the fundamental frequency itself as attenuated by the low-frequency skirt of the filter. If the sweep had been continued to 15 kHz, the right-hand portion would be a tracing of the shape of the bandpass filter from below resonance to the resonance frequency.

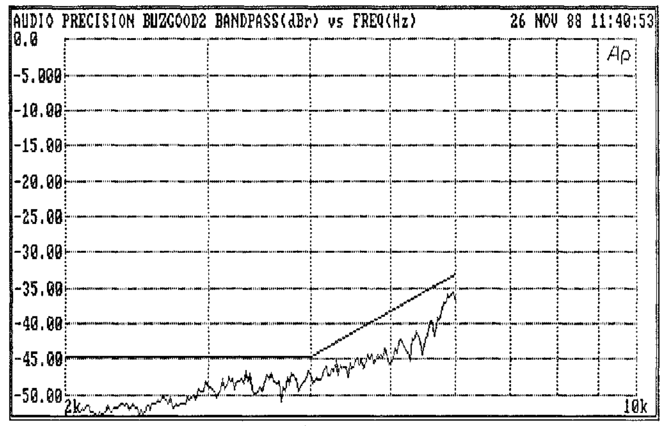


Figure 24 *Rub and Buzz Test Result, Good Tweeter #2*

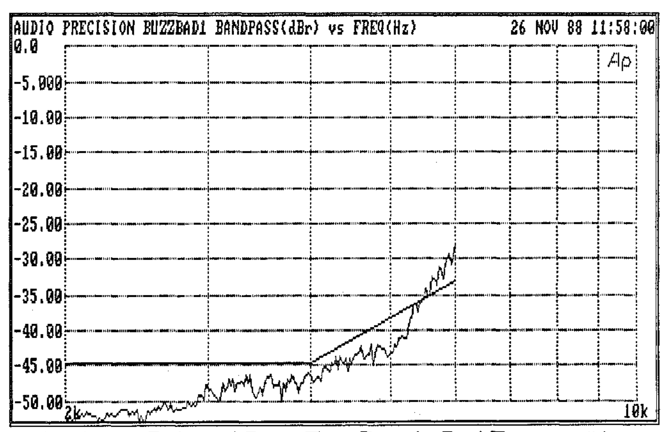


Figure 25 *Rub and Buzz Test Result, Bad Tweeter #1*

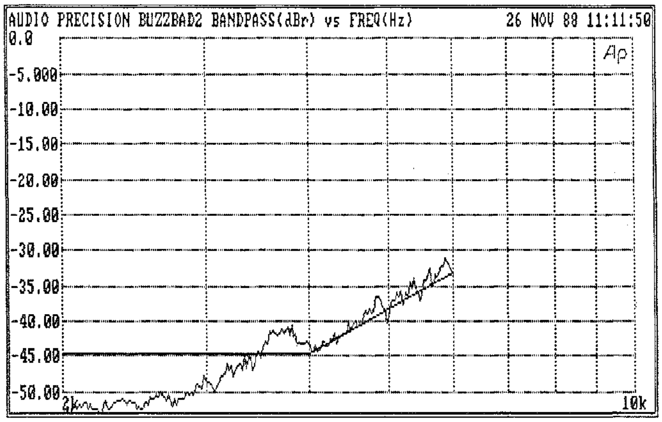


Figure 26 *Rub and Buzz Test Result, Bad Tweeter #2*

Figure 24 is the measurement of another good tweeter. A limit file superimposed on both graphs is horizontal at -45 dB from 2 kHz to 4 kHz, then rising to -33 dB at 6 kHz. Figures 25 and 26 are the measurements on two tweeters with buzz problems. Bad tweeter #1 stayed within limits below about 5.5 kHz, implying that the buzz is either excessive 2nd harmonic distortion or some broadband noise with a portion of the noise spectrum through the 15 kHz region. Bad tweeter #2 shows a relatively abrupt increase in 15 kHz noise output when driven with frequencies between about 3.5 kHz and 3.8 kHz, with marginally-high 15 kHz output at all frequencies above that.

These four illustrations are for rub and buzz testing of a particular tweeter. Testing of a full-range multi-way speaker system is likely to require at least three tests. For example, one test might sweep from 10 Hz to 100 Hz with the bandpass fixed at about 300-600 Hz. A second might sweep from 100 Hz to 1 kHz with the bandpass fixed at perhaps 3-6 kHz. A third test might sweep from 1 kHz to an upper limit somewhere between 5 and 10 kHz, with the bandpass in the 10 to 20 kHz area.

As in the case of most other loudspeaker measurements, acceptance limits for rub & buzz testing are not likely to be set analytically or mathematically. Instead, some highly-qualified listener or listeners will audition a variety of production speakers and subjectively determine which are acceptable and which are not. The test engineer must then experimentally select stimulus amplitudes, frequency sweep ranges, and bandpass filter positioning to yield curves to which limits can be attached for consistent separation of the good and bad units.

Impedance Measurements

Though not typically a part of production test routines, impedance testing can be quickly done by System One in two fashions.

Figure 27 shows the panel setup (stored as IMPEDANC.TST on the companion diskette) and Figure 28 the resulting curves for a full-spectrum sweep of a four-way speaker system while measuring both magnitude and phase of impedance. Figure 29 shows the connection block diagram and Figure 30 a simplified electrical circuit. No power amplifier or microphone is involved; only a two-conductor cable from generator output to speaker system terminals is required. The generator source impedance is selected as 600 Ohms, which is high compared to most loudspeaker systems. Unbalanced mode is selected, which

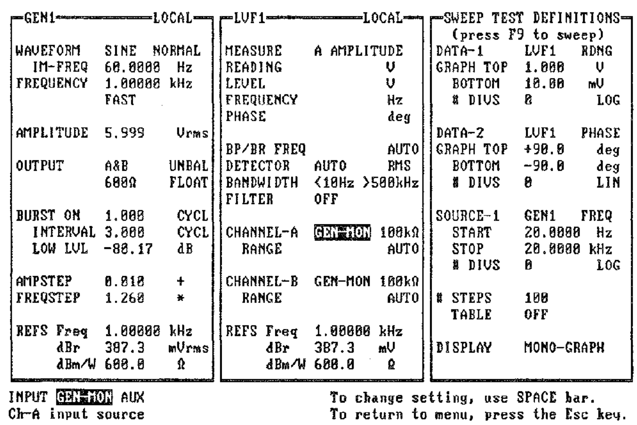


Figure 27 *Setup Panels, Impedance vs Frequency Test. Note Both Generator Outputs ON, Both Analyzer Inputs GEN-MON*

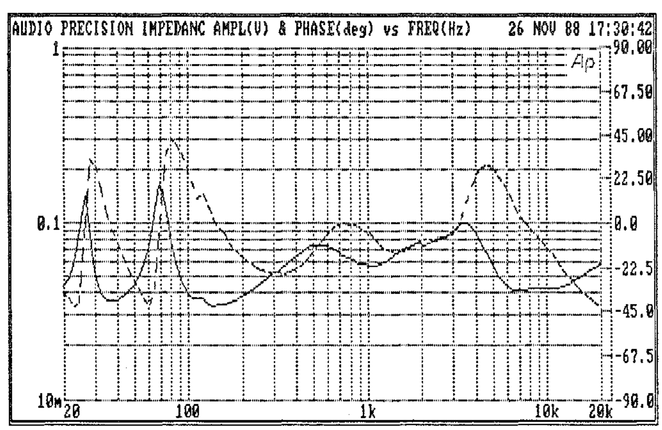


Figure 28 *Magnitude and Phase of Impedance vs Frequency, Four-Way Loudspeaker System*

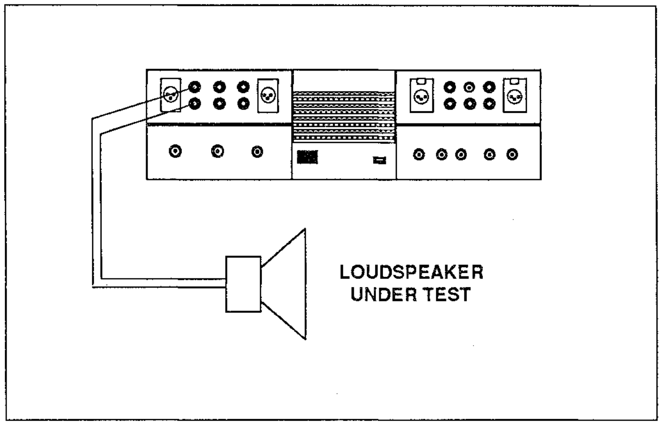


Figure 29 Connection Diagram, Impedance Tests

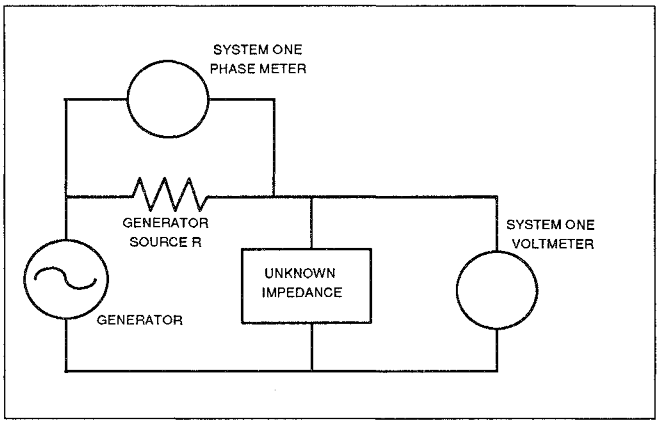


Figure 30 Simplified Equivalent Circuit, Impedance Tests

doubles the generator's maximum current delivery capability. The generator open circuit output amplitude (emf) is set to 6 Volts. A 6 Volt emf behind a source resistance of 600 Ohms results in a constant current source of 10 milliamperes, as long as the external load resistance does not become a significant fraction of 600 Ohms. Both generator output channels are turned on. GEN-MONITOR is selected for both analyzer input channels. The result is that analyzer channel A is measuring voltage across the external load resistance while a (nearly) constant current of 10 milliamps is being forced through the load. The solid (DATA-1) line on the graph is thus the magnitude of loudspeaker impedance, calibrated on the left scale with a factor of 10 millivolts per Ohm. The 0.1 Volts line therefore represents 10 Ohms, and the 10 millivolt line is one Ohm. The phase meter measures phase between the unloaded generator B channel output connector and the A output connector driving the speaker, so the dashed (DATA-2) line is the phase angle of impedance.

Thiele-Small Parameter Testing

A second method of determining loudspeaker input properties is to measure and compute the Thiele-Small parameters. A special purpose program has been written by Dr. Richard Cabot and Robert Wright in Microsoft QuickBASIC, using Audio Precision's LIB-MIX function library to control the instrument and make measurements. This program does not use Audio Precision's S1.EXE software which was used for all the preceding sections of this note, but runs as a compiled (.EXE file) program directly from DOS.

If a DCX-127 is available, the program will automatically measure the dc resistance of the voice coil, make a set of measurements and computations to derive the unloaded parameters of the driver, prompt the user to place the driver onto a sealed box of known volume, and then measure and compute the loaded parameters. If the program senses that the DCX-127 is not present, is prompts the operator to enter the dc resistance value from the keyboard. The program, called DCXSPKRQ.EXE (and the source code listing DCXSPKRQ.BAS) are on the companion diskette to this note. A detailed description of this program can be found in Appendix One of this note.

APPENDIX ONE: AUTOMATED MEASUREMENTS OF LOUDSPEAKER SMALL SIGNAL PARAMETERS

Richard C. Cabot

A computer controlled system for measuring small signal parameters of loudspeakers will be presented. The system uses a signal generator, voltmeter, phasemeter and ohmmeter to measure the free air resonance frequency and Q. The loudspeaker is placed on a box and the measurements are automatically repeated. These measurements, in conjunction with a voice coil resistance measurement, give all of the information needed for calculation of the loudspeaker parameters. The program calculates and prints the values. A flowchart and a listing of the Microsoft QuickBASIC language program are given. Examples of the measurements will be presented.

Introduction

The most common method of measuring small signal parameters of loudspeakers is to measure the loudspeaker impedance both in free air and on a test box of known volume. The resonant frequency and the impedance at resonance are measured. Then the frequencies at which the impedance is 3 dB down from the resonant value are determined. From this data, and the DC resistance of the voice coil, the small signal parameters may be computed. Anyone who has performed this measurement knows how straightforward yet time consuming it can be. Such tasks are ideal for computer automation.

Glossary of Symbols

- f_{CT} resonant frequency of driver in closed test box
- f_S resonant frequency of driver in free air
- Q_{ECT} Q of driver at f_{CT} considering driver electrical resistance R_E only
- Q_{ES} Q of driver at f_S considering driver electrical resistance R_E only
- Q_{MCT} Q of driver at f_{CT} considering driver nonelectrical resistances only
- Q_{MS} Q of driver at f_S considering driver nonelectrical resistances only
- R_E dc resistance of driver voice coil
- r_0 ratio of voice coil maximum impedance $Z(f_S)$ to voice coil resistance R_E
- V_{AS} volume of air having same acoustic compliance as driver suspension
- V_T volume of closed test box
- W_S free air resonant frequency of driver in radians
- $Z(f_S)$ impedance at free air resonance
- $Z(f_{CT})$ impedance at resonant frequency in test box
- $Z(f)$ impedance at frequency f

Test Setup

The equipment for performing the impedance measurements is diagrammed in Figure 1. The hardware consists of an Audio Precision System One audio test system, a DCX-127 digital multimeter, and an IBM PC. Contained in the System One is a sinewave generator, AC voltmeter and phasemeter. The DCX-127 is used for voice coil resistance measurements. The System One operates directly from the PC via an interface card provided with the system. A large sealed box is also required for the loaded measurements.

The loudspeaker under test is driven by the System One generator in the 600 Ohm output mode to approximate a current source drive. The voltage across the loudspeaker under test is measured by channel A of the System One voltmeter, which is also one input of the phasemeter. The

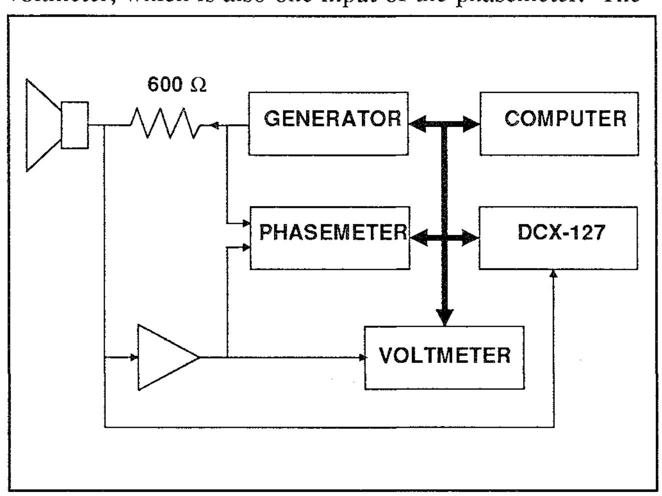


Figure 1 *System Block Diagram*

second channel, which is the phasemeter second input, is connected to monitor the generator open circuit voltage. The DMM is connected across the speaker terminals in parallel with the System One voltmeter input.

Measurements

Running a sweep of frequency and measuring phase and amplitude will give a graph similar to that in Figure 2. Notice that the phase goes through zero at the resonant frequency and changes polarity above and below this point. This fact is important since it greatly simplifies finding the resonant frequency. A simplified version of the impedance magnitude curve is shown in Figure 3 with the important points labeled.

The conventional method of calculating Q is to find the frequencies at which the impedance is 3 dB down from the value at resonance. The resonance data is measured for the free air condition and for the driver mounted on a known volume test box. Small (1972) has derived equations for calculating the driver parameters from these measured values. This technique does not lend itself to automation. To find the resonant frequency requires searching in frequency for a zero degree phase value. Finding the frequencies at which the impedance is -3 dB requires two additional searches, adjusting frequency and measuring voltage. This process is repeated with the driver on the test box, resulting in six search procedures.

A better scheme, after finding the resonant frequency, is to measure the impedance a fixed percentage away in frequency from resonance. By re-deriving Small's equations for the general case of an arbitrary frequency shift we may still calculate the appropriate parameters. This technique requires only two searches and four measurements to obtain the needed data and can be performed much faster than the original approach.

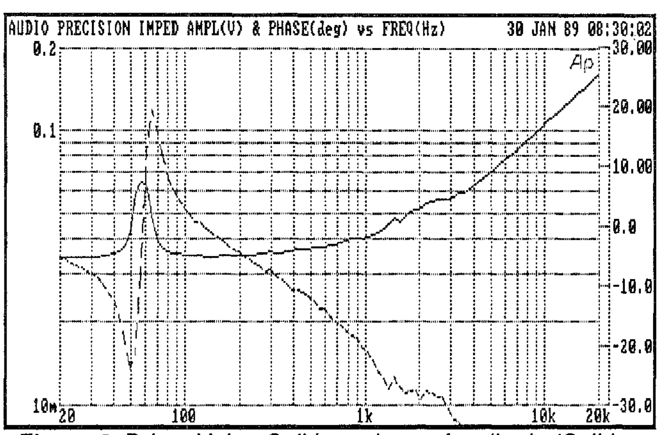


Figure 2 *Driver Voice-Coil Impedance Amplitude (Solid Line) and Phase (Dashed Line) vs Frequency*

The flow chart in Figure 4 shows the basic program flow for automating these measurements. First the DCX is set to ohms and the resistance is measured. The resistance of the voice coil is stored as R_E. The System One is programmed to the initial conditions and a phase measurement is taken. The absolute value of the phase reading is tested to see if it is adequately close to zero. If it is not, the search routine is entered to find the zero phase point.

This routine uses a binary search procedure, where the initial guess and the initial step size for the search are programmed into the program. The sign of the phase reading is checked to see if it is different from the last phase reading. The first time through this loop the previous phase reading is assumed to be zero. If the sign is different this means that the zero phase point has been crossed. If so the step size is cut in half to avoid skipping over the zero point again. If the phase is less than zero the frequency is increased by the step size and the measurement repeated. If the phase is positive the frequency is reduced by the step size and the measurement repeated. Eventually the search will yield a phase measurement below the 0.3 degree tolerance.

The impedance is measured at this frequency which is the free air resonance. The impedance is then measured at 10% above the resonant frequency and compared to the value at resonance. If the ratio is less than 0.8 the measurement is made far enough down the slopes of the

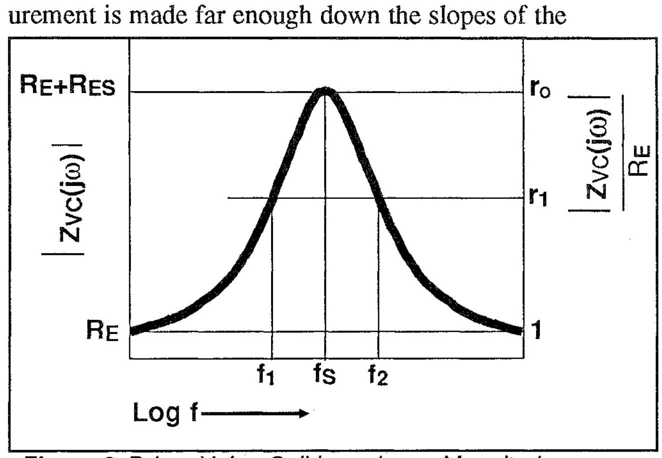


Figure 3 *Driver Voice-Coil Impedance Magnitude*

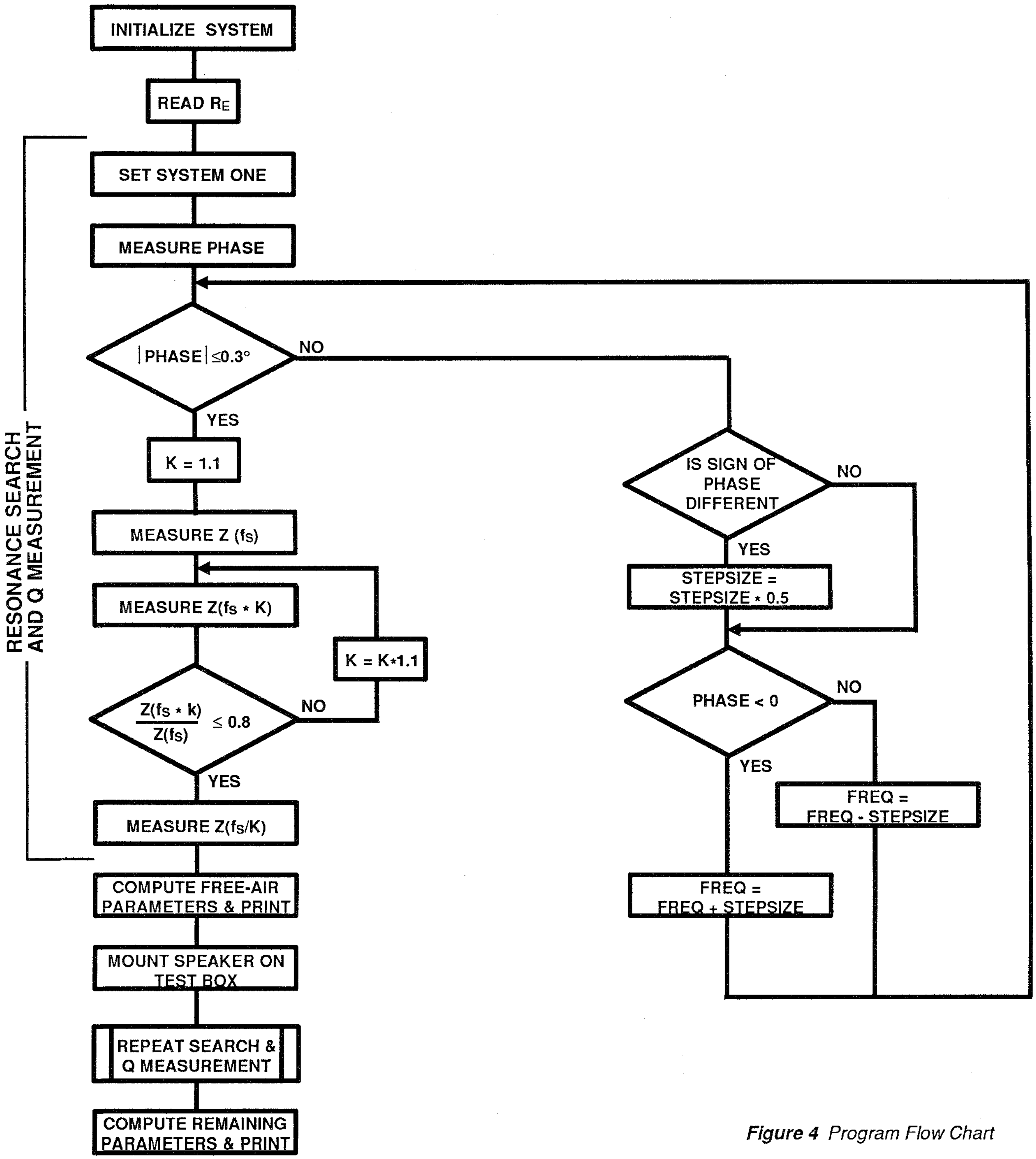


Figure 4 *Program Flow Chart*

resonant peak to yield accurate results. If not, the frequency is incremented by another 10% and the process repeated. When a frequency is found where the impedance is less than 0.8 of the resonant value the frequency is reduced below resonance by the same amount and the impedance is measured. The free air parameters are then computed and printed. The driver is mounted on the test box and the search algorithm is repeated. The new resonant frequency will be the loaded value, f_{CT}. The other loaded driver data is measured and the remaining Thiele-Small parameters are computed.

Theory

Smalls' results will be reviewed here. Equations copied directly from his paper are denoted by an S followed by the appropriate number. We start with Smalls' equation 52 for the mechanical Q in terms of the impedance magnitude at three points of the impedance curve.

$$Q_{MS} = \frac{\omega_s}{\omega_2 - \omega_1} \frac{\sqrt{r_0^2 - r_1^2}}{r_1^2 - 1} \qquad \text{(S52)}$$

we choose the two measurement frequencies on the slopes of the impedance curve a factor of k away from the resonance so that

$$f_H = k * f_s \qquad (1)$$

$$f_L = f_S / k \qquad (2)$$

Substituting these gives

$$Q_{MS} = \frac{1}{k - 1/k} \frac{\sqrt{r_0^2 - r_1^2}}{r_1^2 - 1} \qquad (3)$$

Recall that $r_0 = Z(f_S) / R_E$ and that $r_1 = Z(f_H) / R_E = Z(f_L) / R_E$

Substituting we get

$$Q_{MS} = \frac{1}{k - 1/k} \frac{\sqrt{Z(f_S)^2 - Z(f_H)\,Z(f_L)}}{Z(f_H)\,Z(f_L) - R_E^2} \qquad (4)$$

We now have an equation which relates Q_{MS} in terms of the impedance at resonance and the impedance at two arbitrary frequencies. To maximize accuracy of measurement it is desirable to choose these frequencies close to the -3 dB points of the impedance curve.

The electrical Q and mechanical Q are related by

$$Q_{ES} = Q_{MS} / (r_0 - 1) \qquad \text{(S18)}$$

To compute the driver acoustic compliance V_{AS} the driver is mounted to a closed test box of known volume V_T. The resonant frequency f_{CT} and two offset frequencies f_{HCT} and f_{LCT} are measured as in the free air case. Modifying (4) appropriately we may compute the mechanical Q on the test box Q_{MCT} as

$$Q_{MCT} = \frac{1}{k - 1/k} \frac{\sqrt{Z(f_{CT})^2 - Z(f_{HCT})\,Z(f_{LCT})}}{Z(f_{HCT})\,Z(f_{LCT}) - R_E^2} \qquad (5)$$

The electrical Q is found by

$$Q_{ECT} = Q_{MCT} / (r_0 - 1) \qquad (6)$$

Finally the acoustic compliance V_{as} may be found from

$$V_{AS} = V_T \left[\frac{f_{CT} Q_{ECT}}{f_S Q_{ES}} - 1 \right] \qquad \text{(S19)}$$

Other parameters may be computed as needed using these values.

Conclusions

A basic program has been presented which automates the measurement of loudspeaker small signal parameters using an Audio Precision System One and an Audio Precision DCX-127. For typical woofers the program will return all of the desired parameters in approximately 5 seconds. With minor modifications the system may be used to test drivers on a production line and compare the results to pass fail limits.

References:

Daniels, Drew **Thiele-Small Nuts and Bolts,** AES Preprint #1802, 70th AES convention, 1982

Small, Richard H. **Direct-Radiator Loudspeaker System Analysis**, Journal of the AES, June 1972, Vol. 20 #5, page 383-395

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