Institute of Acoustics

A LOW COST MICROCOMPUTER-BASED INSTRUMENTATION SYSTEM FOR LOUDSPEAKER MEAUREMENTS

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INTRODUCTION

There are three methods commonly used for the measurement of far field loudspeaker response curves:

a) Anechoic room
b) Gated measurement of tone bursts
c) Time Delay Spectrometry (TDS)

Each method has its particular merits but one objective they have in common is to eliminate the distributing effects of nearby reflections. The equipment to be described uses the second method as a compromise between the heavy capital cost and inconvenience of an anechoic room, and the relative complexity of TDS equipment. It consists of a purpose built expansion unit for use with the everyday apparatus of the laboratory; a simple microcomputer, a dot matrix printer and an oscilloscope.

The principle is simple. A variable frequency tone burst emitted from the loudspeaker under measurement is received by a microphone placed, say, one metre away. A delayed measuring gate opens the microphone channel in the centre portion of the tone burst as it passes the microphone, the gate duration and timing being such that the measurement is concluded before the first reflection appears. The sequence is repeated at another frequency when all the reflections have decayed to a low level.

The lowest frequency that can be reliably measured depends on the time the first reflection appears at the microphone; if it is to be 200 Hz no reflecting surface should be within 1.7 metres of the measuring axis, a condition fairly easy to satisfy. The suggestion of 200 Hz as the lowest frequency is based on the normally true assumption that below this frequency the behaviour of loudspeakers is much more predictable and can be better verified accurately by near field or in box (with far field correction) measurements to below 20 Hz. It is quite practicable to obtain an accurate response curve, with a good microphone, over the range 5 Hz to 40 kHz by combining these methods.

EXPANSION UNIT FACILITIES

The expansion unit provides the following facilities:

a) A sine-wave output in 512 steps from 2 Hz to 20 kHz which may be continuous or gated to produce zero crossing start and finish tone bursts of variable duration. Up to 5 watts output into 4 - 8 ohms is available.

b) A separate microphone amplifier with a total gain of 50 dB, switched over a 35 dB range in 5 dB steps. It has a low impedance output and is placed near the microphone so that the rest of the equipment can be kept well away. The gain settings are signalled back to the computer for storage.
c) A peak reading measuring channel (maximum input 1 volt) with a range of 50 dB in 0.2 dB steps or 25 dB in 0.1 dB steps. The channel is gated to allow repetitive measurement of continuous tone, or a measurement of variable duration delayed by a variable time from the start or finish of the transmitted tone burst. The digital results are placed in the computer memory for plotting a response curve on its screen or on paper via its printer. The data may also be stored on magnetic disc or tape.
d) Trigger, measurement gate pulse, and received signal outputs for oscilloscope display.
e) The necessary power supplies including those for capacitor microphone polarising and preamplifier.

All the control signals, except the microphone amplifier gain switching, are handled by the microcomputer. The event timing, being set by the measurement area reverberation, is relatively slow so that the program may be written in a high level language to suit various applications. The only exception is in the tone burst and gate timings which uses a short program written in machine code.

LOGARITHMIC SCALES WITH A-D/D-A CONVERTERS

If logarithmic scaling of linear data is required (as is customary in plotting audio frequency response curves) the calculations may be carried out in the computer from the output of an analogue to digital converter. The capacity of the ADC needed depends on the logarithmic resolution specified and the maximum/minimum ratio of the input. Because the logarithmic resolution varies with the output code (a change from 1 to 2 represents 6 dB resolution, 2 to 3 is 3.52 dB, and n to n+1 is 20*log10((n+1)/n) dB resolution) the lowest usable code (LC) is fixed by the resolution required and (ignoring integer rounding in the converter):

LC = 1/r-1 where r = 10^ (resolution in dB/20)

The highest code will be LC*10^(dB range/20) and the ADC capacity must equal or exceed this value. As an example, for a vertical scale covering a 50 dB range with a resolution of 0.2 dB the highest code would be 43*316.23 = 13598 and a 14-bit ADC is necessary.

Similar reasoning may be used with digital to analogue converters. To produce logarithmically related frequencies from an oscillator whose output is linearly proportional to voltage, each increment of frequency should be a multiplier equal to the required resolution. For example, to cover the range 200 to 20 000 Hz in 256 steps each step would be an increment of (20 000/200^ (1/256) = 1.0182. The lowest usable code would correspond to this ratio, so it would be 56. The highest code would be 56*20 000/200 = 5600 and a 13-bit DAC would be necessary.

It will be seen that the converters are used very inefficiently as codes less than LC cannot be used if the wanted resolution is to be obtained, and most of the codes above LC result in excessive resolution. An alternative is to process the analogue data by a logarithmic or exponential amplifier and then apply it to a relatively inexpensive 8-bit converter. Such amplifiers, using discrete components with three operational amplifiers and a 5-transistor array, include temperature stabilisation and have negligible errors over a 50 dB range. With this approach the data can be stored in 250 steps of 0.2 dB, with enough left over to provide under- and over-range flanges. By electronically switching the logarithmic amplifier to double its output an alternative scale of 25 dB with a resolution of 0.1 dB is available.

The frequency scale can be treated similarly, an 8-bit DAC followed by an exponential amplifier covers two decades of frequency in 255 increments of 1.82% (equivalent to 0.156 dB on the vertical scale).

CONTROL AND DATA TRANSFER

Two addresses are required to operate the unit, both in read and write modes.

Write measurement (WRME) sends a data byte determining the output frequency. Write function (WRFN) sends seven data bits; one bit (TGE) initiates the transmit gate, three bits control the measurement gate and its associated sample and hold circuit, one bit (-SC) starts the measurement analogue to digital conversion, one bit sets the VCO range, and one bit sets the measurement dB range.

Read measurement (RDME) Takes a byte from the measurement DAC and stores it in the microcomputer memory. Read function (RDFN) takes 5 bits; one bit (TG) signals the start and finish positive going zero cross points of the gated transmit signal, one bit (-BUSY) the DAC status, and three bits to store the settings of the microphone amplifier gain switches.

Because a high level language program may be too slow to time the gates accurately, the CPU may be accessed directly with a machine code routine. The microcomputer used for this unit was Z80 based running at 2 MHz. The BC, DE and HL registers are loaded with the transmit duration, measurement delay and measurement delay + duration data. A signal is sent to initiate the transmit gate and when the next oscillator positive going zero crossover occurs the transmit gate opens and starts the timers. Each register is decremented in turn until it reaches zero. BC = 0 initiates the transmit gate termination which closes on the next positive going crossover. DE = 0 opens the measurement gate and HL = 0 closes the measurement gate. The digital to analogue routine is now started which sets or resets the bits controlling the hold capacitor switch and the DAC. When the DAC BUSY signal ends the hold capacitor is discharged for a period loaded into the HL register and the CPU returns to the high level program control.

TRANSMITTER CHANNEL

The DAC output, via an exponential amplifier, drives an 8038 VCO which produces a sine-wave signal that can be varied over a frequency range of two decades. The tuning capacitor is selected by an FET switch; the range 200 to 20 000 Hz is primarily used for gated measurements and the range 2 to 200 Hz is intended for near field or in box measurements. Distortion from the VCO is typically -45 dB which should not cause any significant measurement errors, but it is of a high harmonic order which sounds unpleasant and cannot be used for listening checks on a loudspeaker. Apart from driving the small power amplifier, via a CMOS gate switch, the sine wave is squared and used to clock a positive edge triggered D-type flip-flop which generates the transmit gate pulse.

The power amplifier output is metered during set-up by connecting it through a fixed attenuator to the receiver channel gated rectifier.

RECEIVER CHANNEL

The receiver channel is intended for use with a standard 1/2" capacitor microphone and preamplifier of about 12.5 mV/Pa sensitivity. Lower grade microphones may be used and corrected with a software look-up table for production testing of loudspeakers.

A typical loudspeaker will generate a pressure of 0.5 Pa (+88 dB to 20 μPa) at a distance of 1.0 m for 1 watt electrical input, so that the microphone output will be about 6 mV. The gated signal rectifier requires 1V to produce its full output of 7.07 V (operation at a high level yields the greatest linearity over the widest bandwidth) so that a gain of +45 dB is needed to fully load the rectifier. This is effected by four amplifiers in series. The first has an adjustable gain of between 8 and 15 dB which is set during the system calibration. The other three have a gain switchable between 0/+20 dB, 0/+10 dB and 0/+5 dB respectively. Thus the additional gain can be varied from 0 to +35 dB in 5 dB steps. The switches also set a bit in the RDFN latch so that the microphone amplifier gain can be stored with the measurement data to produce the correct dB scale in the frequency response display.

A full wave rectifier system is employed which is substantially linear over a range of 50 dB up to 20 kHz. During the measurement gate duration a capacitor is charged to the peak positive or negative part of the signal. The charge is then held, logarithmically converted, and applied to the DAC which is enabled by -SC. At the end of the DAC cycle the capacitor is discharged to await the next measurement.

OPERATION

The transmit gate pulse is used to trigger the time base of a dual trace oscilloscope. The measuring gate is displayed on one trace and the received signal on the other as an aid to the setting up of the system. It is also instructive sometimes to observe what the loudspeaker makes of a tone burst.

Although the tone burst starts and ends at the zero crossover point of the sine wave, starting and stopping transients are present. The unit under test will also have a finite bandwidth and the burst duration should be sufficient to ensure the transducer output has reached a steady condition. For a lower frequency limit of 200 Hz the shortest burst duration will be 5 ms in order to accommodate one complete sine wave. The measuring gate should be shorter thant this to eliminate the starting and finishing transients but should, at least, encompass the positive and negative peaks of the sine wave. This suggests that the minimum measuring gate should be 2.5 ms, starting 1.25 ms after the start of the tone burst has been received. The measuring gate has to be delayed additionally by the time taken for the tone burst to reach the microphone (2.9 ms/m), and also by the group delay of the transducer. In practice the setting up is straightforward. The burst length is set to about 5 ms, the measuring gate to about 3 ms and the dalay adjusted so that the measuring gate is symmetrically within the received burst as displayed on the dual trace oscilloscope. The inter burst delay (typicaly 125 ms) depends on the reverberation time of the measuring area and is set so that the previous burst has decayed by at least 30 dB. This may be checked by reducing the measurement gate and delay so that it ends just before the tone burst is received. These parameters can be stored in the software so that the system automatically runs with suitable settings with an option to vary them if needed.

CONCLUSION

The advantages of using a general purpose small computer running a high level language with purpose built hardware are considerable as it is very easy to write special programs for special purposes. Some examples follow:

a) Comparison curves can be produced showing, for example, the effect of a grille over a loudspeaker by storing before and after data, then plotting a difference curve.

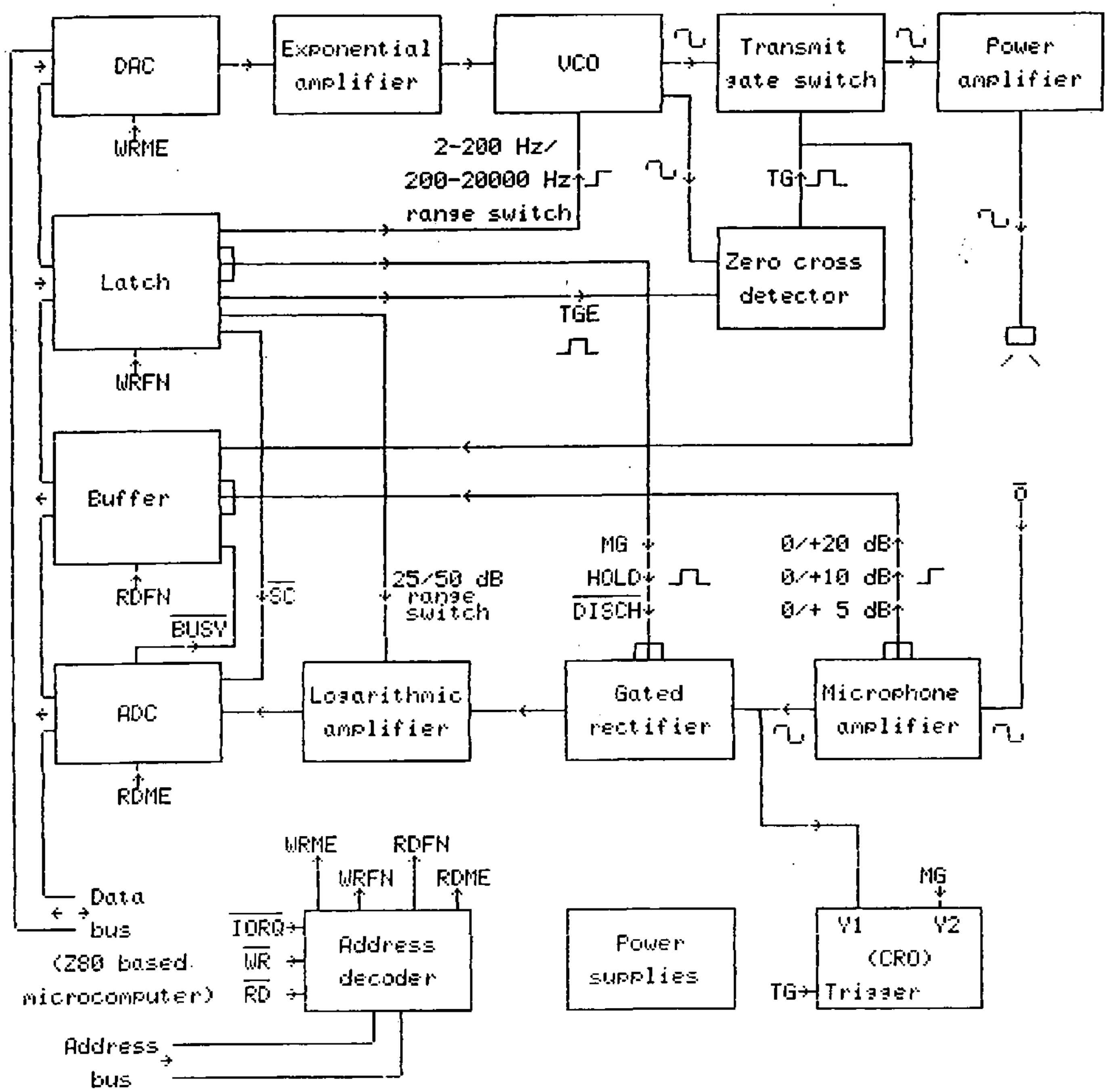
b) Microphone responses can be plotted by measuring a (preferably good) loudspeaker with a standard microphone, remeasuring it with the microphone under test in the identical position, then plotting a difference curve.

c) Microphone responses may be corrected by providing a look up table which modifies the measured data.

d) Composite response curves can be plotted comprising loudspeaker near field and gated responses in which the change over points are matched.

e) Go/no go production tests can be carried out by comparison with stored data from a reference transducer.

f) By timing the measurement delay from the end of the tone burst, decay curves can be produced (provided that nearby reflections are checked).


DAC
Exponential amplifier
VCO
Transmit gate switch
Power amplifier
WRME
2-200 Hz/
200-20000 Hz
range switch
TG
Zero cross detector
Latch
TGE
WRFN
Buffer
MG
HOLD
DISCH
0/+20 dB
0/+10 dB
0/+ 5 dB
RDFN
BUSY
SC
25/50 dB range switch
ADC
Logarithmic amplifier
Gated rectifier
Microphone amplifier
RDME
WRME
RDFN
WRFN
RDME
Data bus (Z80 based microcomputer)
IORQ
WR
RD
Address decoder
Address bus
Power supplies
MG
Y1
Y2
(CRO)
TG
Trigger