

THE SUBSONIC BAND IN COMPUTER SYSTEMS BASED ON A SOUND CARD

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UDC 621.317.3:004.42:534.6:534-6

A circuit for improving the sound card for use in a computer multifunctional spectrum analyzer and signal generator with a frequency range extended into the subsonic region is proposed.

Key words: *subsonics, sound card, generator, spectrum analyzer.*

The use in computer devices – a multifunctional spectrum analyzer [1] and a signal generator [2] – as analog-to-digital and digital-to-analog converters, of a computer sound card enables the problem of investigating audio-band signals to be solved extremely economically. However, the extension of the frequency band with respect to the electric voltage into the subsonic and dc band gives rise to certain difficulties. Many widely used computer sound cards have a low-frequency limit of about 10–20 Hz. This limitation is due, in the majority of cases, to the presence of isolating capacitors at the input of the analog-to-digital converter and at the output of the digital-to-analog converter. Unfortunately, a simple removal of the isolating capacitors may often be insufficient for operation in this range. This is due to the fact that, as a rule, both the input of the analog-to-digital converter and the output of the digital-to-analog converter have a dc bias with respect to the common lead of the analog circuits. The latter is due, in turn, to the fact that the supply of these codecs (the coder-decoder, the digital-to-analog converter and the analog-to-digital converter on a single crystal) are most often of all unipolar (usually 5 V). Consequently, in order to ensure that the sound card operates in the subsonic and dc range, it is necessary to use an additional level-shift circuit both for the input and output signals. Such a solution is essential, firstly, because it is much cheaper to use specialized analog-to-digital and digital-to-analog converters, and it also ensures extremely high parameters. Secondly, this approach enables one to use already existing software [1, 2] to construct measuring-computing systems with a frequency band which extends into the subsonic range.

As an example we will present the solution of this problem for the two-channel 16-bit *CS4232* codec made by the Crystal Semiconductor Corporation, installed in the *TBS2000* sound card manufactured by the Turtle Beach Systems Company.

The basic circuit of the zero-level shift unit for the input and output signals together with a fragment of the codec circuit is shown in the figure. We show the names and numbers of the units in the *CS4232-KQ* microcircuit enclosed in the *TQFP* 100-contact housing. All the connections with the *LAUX1* and *RAUX1* leads on the sound card must be disconnected. All must be disconnected from the outlets of *LOUT* and *ROUT*, except the filtering capacitors *C1* and *C2*, shown in the figure.

A reference-voltage source *VREF*, built in to the *CS4232* codec, is used as the reference voltage for the operational amplifiers of the level-shift circuit. The technical documentation on the microcircuit indicates that the input and output voltages of the codec are centered with respect to the reference voltage *VREF*, having a typical value of 2.2 V. Since this source does not allow of any considerable dc load, a voltage follower (the *DA5* microcircuit) is connected to its output. The use of a reference-voltage source built in to the codec considerably reduces the drift of the whole level-shift circuit when the temperature and supply voltage change. The reference-voltage inverters necessary for the zero-level shift of the input signals to operate take the form of the *DA3* and *DA4* microcircuits. The use of separate inverters is dictated by the need to adjust the

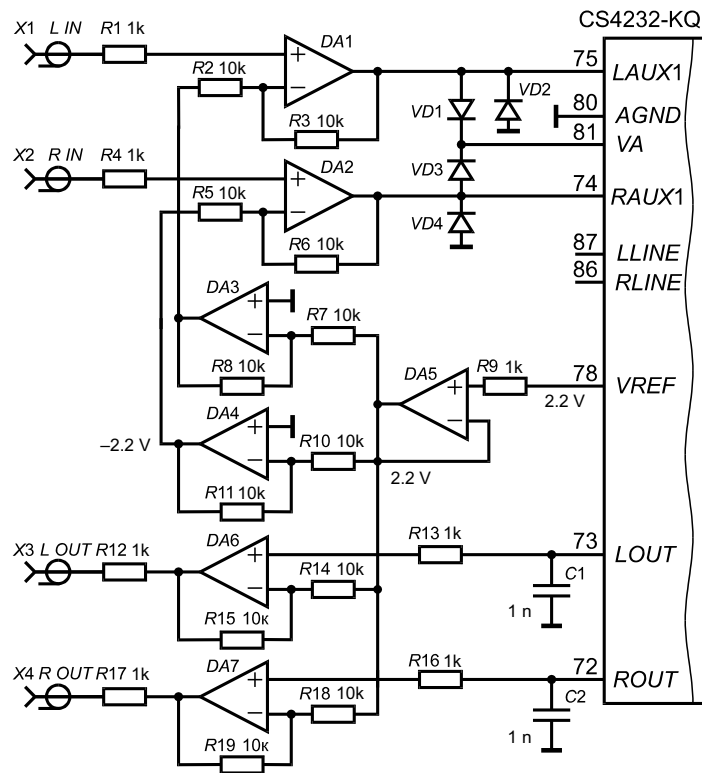


Fig. 1. Basic circuit of the zero-level shift unit of the output and input signals for the CS4232 codec: DA1–DA7 are AD743 or AD745 operational amplifiers, and VD1–VD4 are KD503 diodes.

zero of each of the input channels as accurately as possible, since these, as a rule, have different inherent values of the zero shift. Accurate adjustment of the zero individually for each of the two channels is achieved by choosing the resistances of resistors $R7$ and $R10$.

The zero-level shift unit for the input signals is constructed using the operational amplifiers DA1 and DA2. They serve to subtract the inverted voltage of the reference source from the input (measured) voltage. Hence, the input (measured) signal and the voltage of the reference source are summed. The diodes VD1–VD4 at the outputs of operational amplifiers DA1 and DA2 protect the inputs of the analog-to-digital converter from a voltage higher than the permissible value, which may occur if it is overloaded by input signals or during transients when the whole system is switched on and switched off.

In addition to the inputs LAUX1 and RAUX1, to introduce the signal into the codec one can use the inputs LLINE and RLINE. The input from any one of these is switched in the program mixer, which forms part of the Windows operating system. If all these inputs of the codec are provided with a zero-shift circuit, this enables the input to be switched by a program from the two built-in signal sources without using additional equipment. In this case, the microcircuits DA1–DA4 and also the corresponding resistors and diodes VD1–VD4 must be doubled for the zero shift of the additional channels LLINE and RLINE.

The zero-level shift unit for the output signals of the codec is constructed using the operational amplifiers DA6 and DA7. They subtract the voltage of the reference source from the output voltage of the digital-to-analog converter. Accurate adjustment of the zero is achieved separately for each channel by choosing the resistances $R14$ and $R18$. The capacitors C1 and C2 occur in the construction of the sound card.

To realize the possibilities of the codec completely (a dynamic range of greater than 80 dB), it is necessary to use low-noise devices with field-effect transistors at the inputs and a rate of rise of the output voltage under unit-amplification

conditions of not less than 1 V/ μ sec as the operational amplifiers *DA1–DA7*. The imported AD743 and AD745 low-noise operational amplifiers give different results. The Russian 574UD1 or 544UD2 microcircuits have somewhat worse parameters. The leads of the operational amplifiers are not numbered in the circuit since they differ for the AD743 and AD745 microcircuits, which are manufactured in different casings.

The operational amplifiers with a large zero shift (574UD1 and 544UD2) must be supplied with typical zero-correction circuits, using the circuits usually employed in the reference technical documentation. These correction circuits also enable one to compensate the zero shift of the digital-to-analog converter and analog-to-digital converter of the codec more accurately. If the operational amplifiers used for stable operation require external circuits for correcting the frequency characteristic, they must also satisfy the corresponding technical documentation.

To supply all the operational amplifiers, it is necessary to use a bipolar stabilized low-noise dc voltage source of $\pm(5-12)$ V. Blocking capacitors with capacitances of 0.1 μ F (ceramic or film) and 10.0 μ F (electrolytic) must also be connected in the supply circuits. If additional interference filtering is necessary one can use the voltage source of ± 12 V which already exists in the computer. All the “ground” leads, shown in the level-shift circuit, must be connected with the similar ground of the sound card. In the *CS4232* microcircuit, this contact is *AGND* (lead 80). The last of the *CS4232* contacts – the lead *VA* (81) – is the supply of the analog part of the codec, which is used to connect the protective diodes *VD2* and *VD3*.

The resistors employed in the system must necessarily be low-noise resistors, for example, S2-1 metal-film resistors or similar. Resistors with a nominal resistance of 10 k Ω must be chosen in pairs for each of the operational amplifiers with minimum spread. This is required for the most accurate compensation of the level shift. The nominal resistance of these resistors is not important and can be in the range 5–10 k Ω . Resistors with a nominal resistance of 1 k Ω perform protective functions. Their accuracy is not important.

One can use any high-frequency silicon diodes with low capacitance, for example, the type KD503, as the diodes *VD1–VD4*.

The best construction of the level-shift unit is to make it using a separate screened printed circuit, placed over the sound card. This printed circuit must be provided with a typical metal fastening bracket, on which the output and input coaxial joints are placed.

Additional adjustment of the system to increase its operating accuracy consists of choosing the resistors *R7*, *R10*, *R14* and *R18*. The shift of the input signals is regulated by choosing the resistors *R7* and *R10* when connected with the common lead of the analog input circuits of the system (joints *X1* and *X2*). The computer oscilloscope should serve as an indicator of the adjustment [1].

The shift of the output-signal amplifiers of digital-to-analog converters *DA6* and *DA7* is adjusted by choosing the resistors *R14* and *R18*. A signal of zero amplitude must be fed into both channels from the generator [2], for which one presses the key marked Mute. One must use a dc millivoltmeter or the already adjusted input part of the system described as a null indicator.

Other sound cards constructed using digital-to-analog and analog-to-digital converter microcircuits having a structure similar to the *CS4232*, for example, the *CS4235* and *CS4197*, can be subjected to the same alterations. Obviously the numbering of the leads and their formal names may differ from those shown in the figure. These data can be obtained in the technical documentation of the specific microcircuits. Such information, as a rule, is at present freely available on the internet-sites of companies which produce digital-to-analog and analog-to-digital converter microcircuits. Whereas digital-to-analog and analog-to-digital circuits, on separate microcircuits, are used on the sound card, appropriate leads must be used as sources of reference voltage for the level-shift circuits of the input and output signals.

It is absolutely necessary here to make one important observation. In high-quality (and fairly expensive) sound cards, high-frequency digital filters are often used having a cutoff frequency of the order of several hertz. This applies, for example, to the FIJI sound card made by Turtle Beach Systems. Such filters are designed to compensate shift and drift of the zero of the input analog-to-digital converter. The filters can be on a crystal of the codec or built in to the digital processor for processing the signals, or have a program form in the driver. In any of the above cases, this makes it impossible to use such a sound card for inputting very low frequencies. This can only be ascertained before starting the adjustment with some degree of probability. To do this, it is necessary to estimate the zero shift and drift of the analog-to-digital converter when

there is no input signal (the input is grounded) using an oscilloscope, built in to the combined computer spectrum analyzer [1]. If the measured shift exceeds 5–10 units of the lowest digit, possibly already changed due to initial heating, then almost certainly there is no digital filter. If the shift is considerably less this indicates that either a very good and stable analog-to-digital converter is being used, or a high-pass digital filter is employed.

One other method for making a preliminary check of the actual possibility of inputting a constant voltage is to make a connection between its output *ROUT* and the input *RAUX1* (in terms of the *CS4232* codec) and to measure the amplitude-frequency characteristic in the extremely low-frequency region (down to 10^{-3} Hz) using a generator [2] and an oscilloscope, forming part of the computer spectrum analyzer [1]. This method of checking ensures a reliable result in practice, but requires preliminary interference (although only a small amount) in the sound-card circuit.

Lead out of the subsonic signal and the direct current using the method described in this paper is usually possible without any difficulties for any sound card irrespective of the type of digital-to-analog circuit employed.

The sound card developed in this way operates at a lower working frequency of 10^{-2} Hz when using the combined computerized spectrum analyzer [1], and its lower limit will be a constant current when it includes an oscilloscope. For a computer signal generator [2] the lower limit of the frequency band with respect to the output electric voltage is 10^{-3} Hz. The upper limit of the working frequencies is determined by the maximum digitization frequency of the *CS4232* codec and reaches 20 kHz.

The nominal input and output voltages, corresponding to the complete scale of the analog-to-digital and digital-to-analog converters, are ± 1.4 V. The highest permissible input voltage for the device described depends on the parameters of the operational amplifiers employed and in any case should not exceed the supply voltage of the latter in absolute value. The input resistance is greater than 1 M Ω and the output resistance is less than 200 Ω .

The residual zero shift of the converters after compensation is determined by their internal errors and most often does not exceed 0.05% of the total scale for the digital-to-analog converter and 0.02% for the analog-to-digital converter. Final compensation of the residual zero shift of the analog-to-digital converter is achieved by programming the spectrum analyzer [1]. If higher requirements are imposed on the stability of the zero of the converters it is advisable to keep the sound card at a constant temperature in a thermostat.

The dynamic range of the sound card described above, measured when the output is connected to the input, using the low-noise AD743 or AD745 operational amplifiers in the zero-level shift circuit and the optimum structural form, reaches 78–80 dB.

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