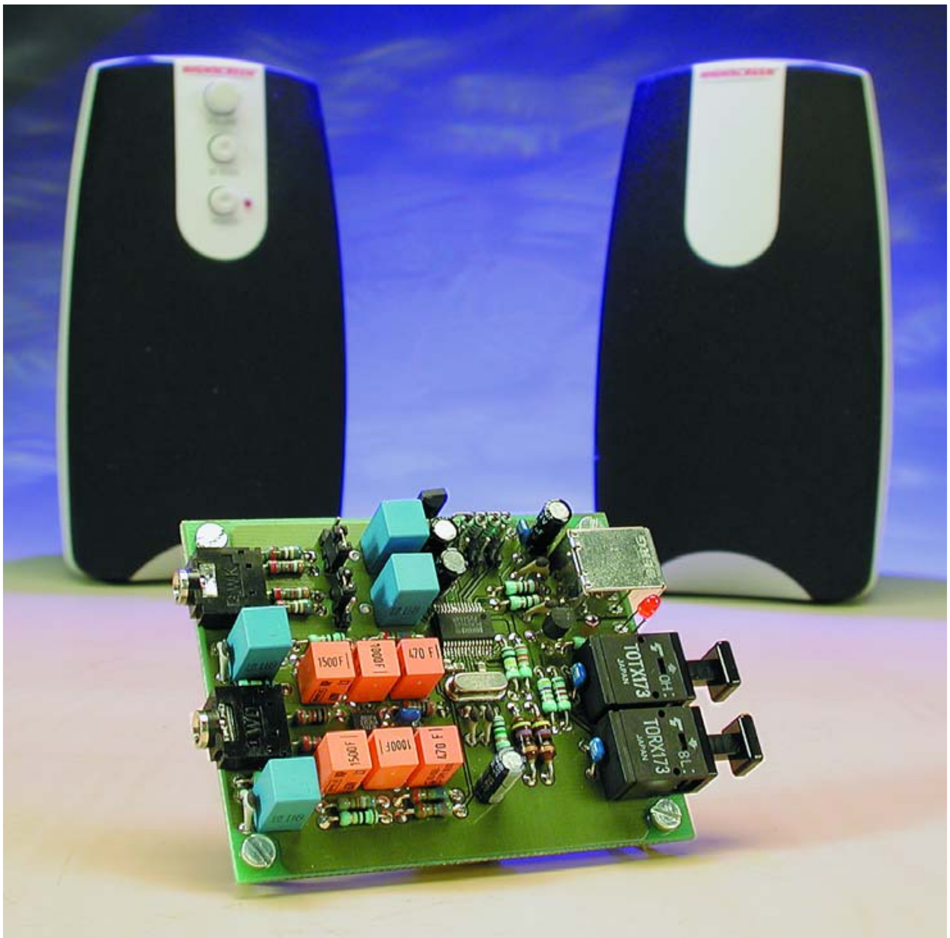
USB Audio Codec with S/PDIF

record digital audio with your PC

Design by T. Giesberts

The USB Audio-DAC described exactly two years ago received unexpected acclaim as a small external sound card. Soon after the publication we received an number of requests for a similar circuit, but then capable of recording via the USB port. The codec presented in this article does just that, and more, while not a much more elaborate circuit than the Audio-DAC.



The circuit described in this article is built around a 'stereo audio codec' IC type PCM2902 from Texas Instruments/Burr-Brown. Clearly, this chip is from the same family as the PCM2702 we used in the USB Audio-DAC. However, it is also an upgrade because besides the features of the 2702 (analogue stereo output, USB interface) it also sports an analogue stereo output and an S/PDIF input and output. The latter even allow digital recordings to be made to and from the PC. One proviso should be mentioned, however. The PCM2902 processes digital audio signals according to the Serial Copy Management System (SCMS) and will switch to the analogue input if it discovers a copied signal. If signals are simultaneously applied to the S/PDIF and analogue inputs, the digital input is automatically selected (if original data is being applied).

If you want to know all the ins and outs of this interesting IC, get the datasheet from www.ti.com. To wet your appetite the block diagram of the PCM2902 is given in **Figure 1**.

Just as with the PCM2702, the 2902 guarantees a very decent quality of the signal processing, so there's no need for audio purist to skip these pages! Those of you who insists on performance figures to be convinced of the need to build a circuit are referred to the inset which lists some measurement results obtained from our prototype.

KISS

Despite the fact that we're dealing with some pretty complex circuitry hidden away in the PCM2902, the final design meets the KISS requirement (keep it simple stupid). Besides the audio codec, IC1, the circuit has just a few more active parts — a transistor, a voltage regulator, a double opamp and an optical input and output module. Add a handful of passive parts and Bob's your uncle.

The codec card is bus powered, which means that the entire circuit is supplied from the 5-volt pin of the USB port. Components L1 and L2 are purposely located close to the USB connector to eliminate noise directly at the source. Hence, these inductors are SMDs (surface mount devices) and fitted on the pins of K3, at the underside of the board — more about this further on. Incidentally, the PCM2902 has an on-chip voltage regulator.

Input circuit

The analogue input has a fixed sensitivity, so you will not see the usual slider pot in the window containing the recorder properties. Because of this, each analogue input channel has a jumper for HI/LO (high/low) input level selection. The HI level is set to about 2 V by means of two potential dividers, R1/R2 and R3/R4. The input range of the ADC inside the PCM2902 is determined by the level of VCCCI (0.6 times VCCCI). Every user is, of course, free to replace the fixed potential dividers with a stereo potentiometer. Capacitors C2 and C3 decouple the direct voltage at the IC inputs. The voltage amounts to (approximately) VCCCI/2 (= VCOM) so will be around 1.65 V.

In order to increase the quality of the A-to-D conversion, the manufacturer recommends powering VCCCI by a separate voltage regulator. Hence the addition of IC5 to the circuit. Here, a low-drop 3.3-volt regu-

PCM2902 FUNCTIONAL BLOCK DIAGRAM

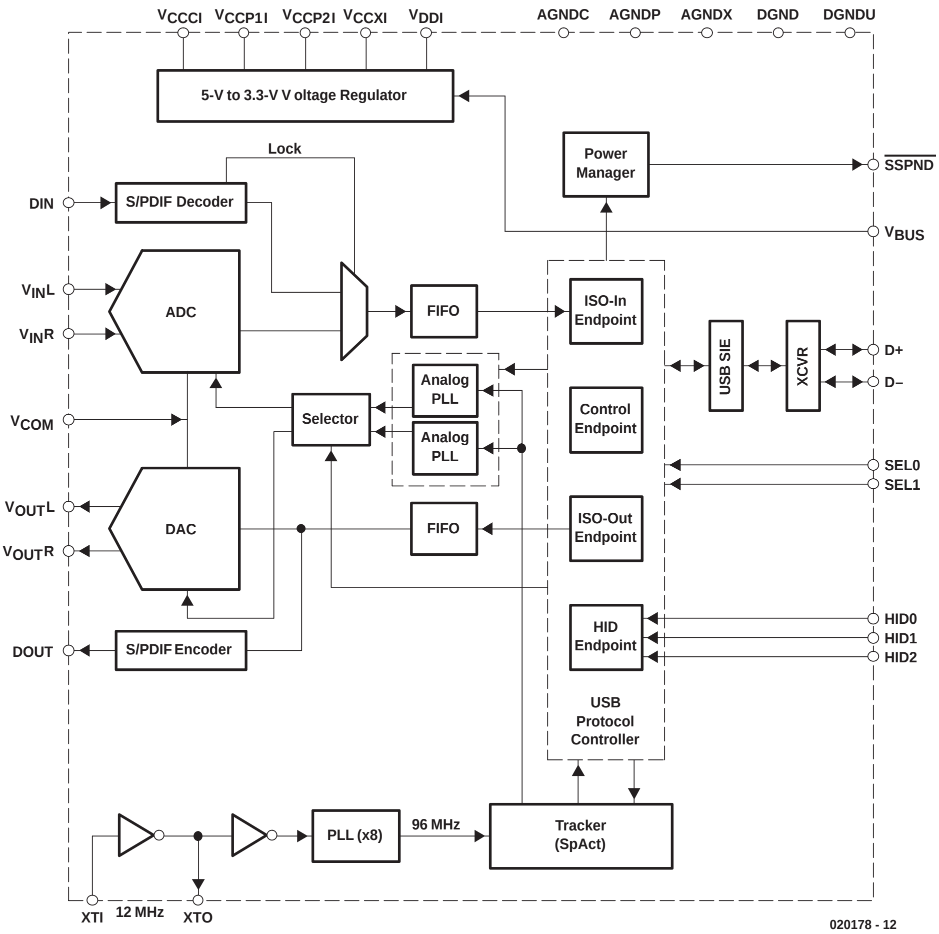


Figure 1. Internal architecture of the PCM2902 Stereo Audio Codec.

Measurement results

(VBUS= 4.84 V, VCCCI = 3.5 V)

Current consumption	approx. 90 mA
DAC	
Nominal output voltage (0 dB)	1.1 V_{rms}
Frequency range (–3 dB)	22.7 kHz (f_s = 48 kHz)
Relative amplitude at 20 kHz	–0.8 dB (f_s = 48 kHz)
Analogue filter bandwidth	28 kHz
Output impedance	100 Ω
Signal/noise ratio	> 95 dBA
THD+N (1 kHz)	0.005% (B = 22 kHz)
	0.046% (B = 80 kHz)
Channel separation	> 99 dB (1 kHz)
	> 76 dB (20 kHz)
ADC	
Input voltage range	0.75 V_{rms} (LO)
	2.1 V_{rms} (HI)
Input impedance	7.7 kΩ (LO)
	10 kΩ (HI)
THD+N (1 kHz, –0.5 dBFS)	0.01% (B = 22 kHz)
Channel separation	> 73 dB (1 kHz)
	> 47 dB (20 kHz)

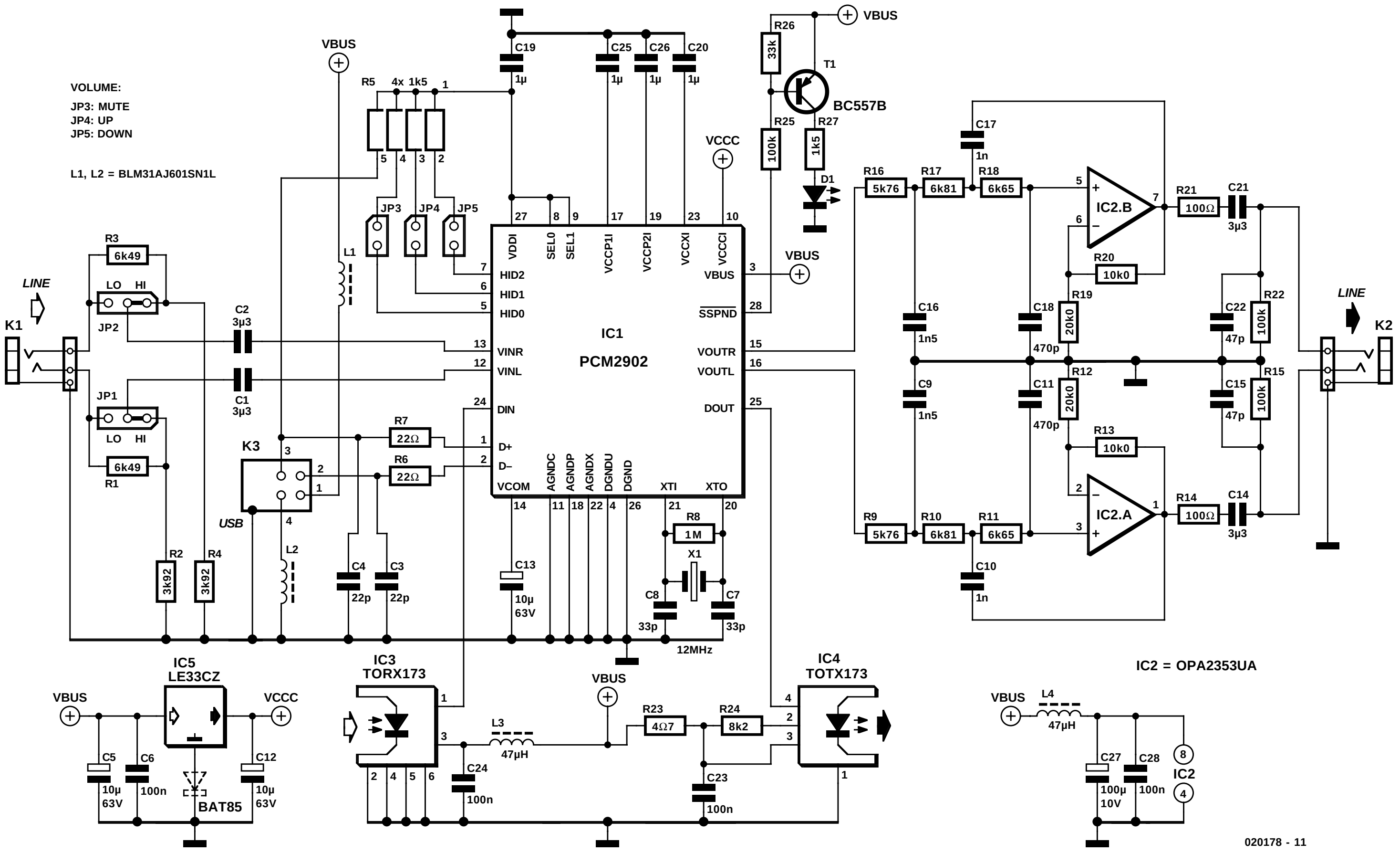


Figure 2. The circuit diagram excels in simplicity because all the complex stuff is inside IC1.

lator type LE33CZ is used. The IC is supplied by ST and comes in a TO92 case. Its presence reduces harmonic levels by more than 10 dB as compared with the internal voltage regulator. If you want to squeeze the ADC for lowest distortion, you may insert a BAT85 diode in the ground line (centre pin) of the LE33CZ. This optional addition is shown in dashed outline in the circuit diagram. The anode of the BAT85 diode goes to the regulator ground pin. The diode raises the reference voltage from 3.3 to about 3.5 V. Although a dedicated 3.5-V regulator does exist under the designation LE35CZ, this component is not easily available.

Because the ADC has an on-chip anti-aliasing filter, the input circuit is marked by delightful simplicity. However, to ensure good sonic quality, MKT (Siemens) or equivalent capacitors are specified for positions C1 and C2. Do not be tempted to use electrolytics here.

Output circuit

As regards structure and specifications, the DAC closely resembles the 'old' PCM2702. Internally there's a low-pass filter with a roll-off at 250 kHz. As with all delta-sigma DACs, the noise level rises considerably beyond 20 kHz or so due to noise shaping. To eliminate the various distortion products, a third-order Butterworth low-pass filter with f_c = 28 kHz is applied. Because the amplifier for this filter is supplied from the VBUS 5-volt line, the signal at the DAC output may be amplified a little (1.5 times). This ensures a maximum output level of more than 1 V_{rms}. The amplifier is realised by an opamp type OPA2553UA which comes in an SMD SO-8 case. The same chip was applied in the USB Audio-DAC with excellent results. It is marked by high speed, low noise, low distortion and (very important) rail-to-rail voltage swing at the input and output.

The filter outputs are equipped with the standard decoupling capacitors (again MKT types) and small series resistors to cancel the effect of capacitive loads. R15 and R22 ensure that these output capacitors remain charged (the input p.d.'s at the ADC have the same function). C15 and C22 suppress any RF noise (remember, the complete circuit s electrically connected to the PC…). In addition to these measures, the opamp is provided with its own supply decoupling network consisting of L4, C27 and C28.

Other details

We can be brief about the S/PDIF input and output. As illustrated by the circuit diagram, these consist essentially of the well-known Toslink modules — a TORX173 for the input and a TOTX173 for the output. The Toslink modules (IC3 and IC4) are directly connected to the relevant pins of IC1 and they are powered from the +5-volt rail using traditional decoupling components.

The PCM2902 has a HID volume control feature and a mute function. Jumpers JP3, JP4 and JP5 allow mute, volume-up and volume-down to be adjusted respectively. You could, for example, connect three pushbuttons to these jumper pins and so enable basic volume control without access to the PC. When connecting such pushbuttons, be careful to avoid short-circuits to ground.

Installation of this USB component under Windows 98 or higher results in two items being added to the Human Interfaces. Consequently a report is filed back to the operating system specifying the current volume, mute state, etc.

The state of the Suspend flag is reported on pin 28 and an LED is used to indicate that the IC is 'temporarily out of order'. LED is purposely connected through to ground to prevent damage to the IC when a short-circuit occurs. A small converter built around T1 converts the 0 V/3.3 V supplied by the SSPND output into a 2-mA current required for the high-efficiency LED. In this way, the LED connection may be employed to signal to other digital equipment that the signal is temporarily not available.

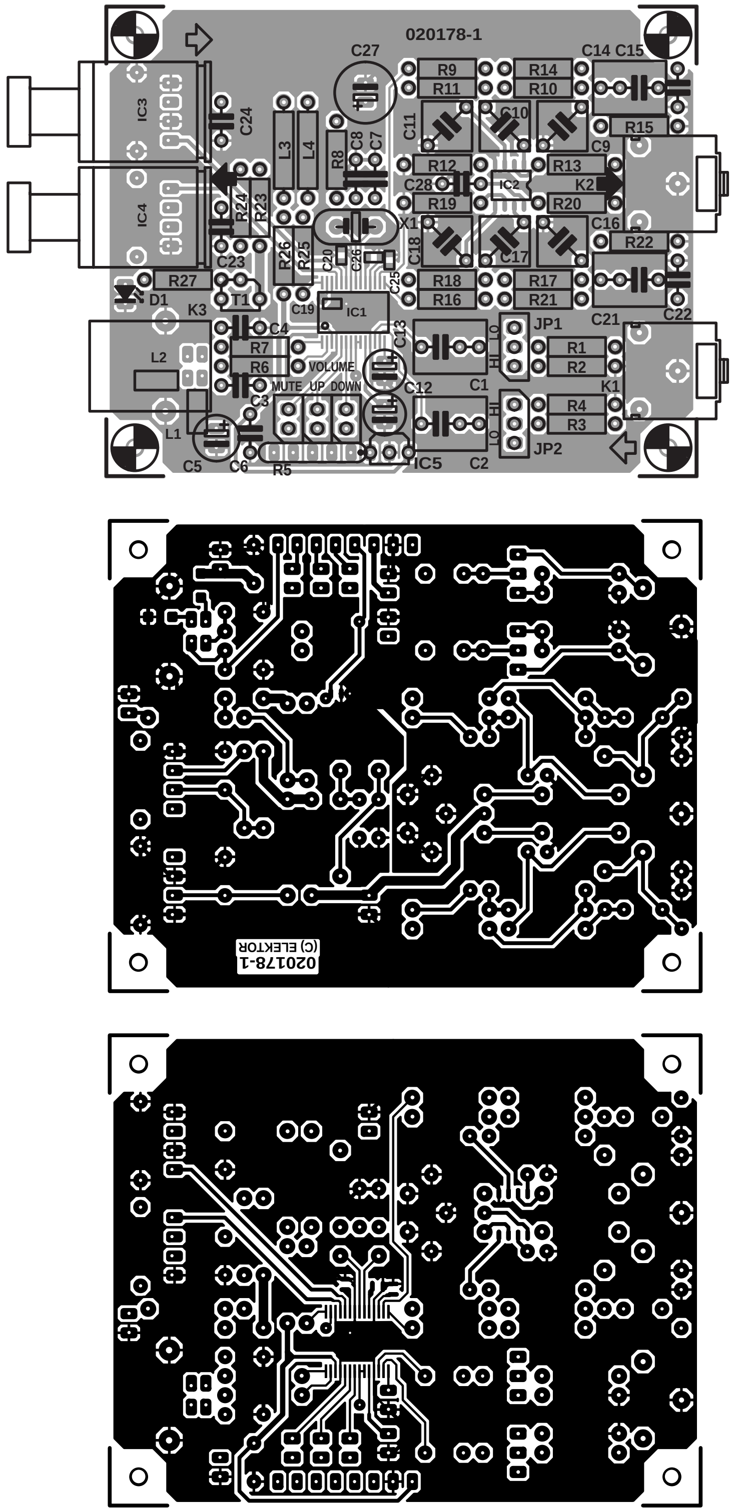


Figure 3. Copper track layout and component mounting plan of the PCB designed for the USB Audio Codec (board available ready-made).

Printed circuit board

The design of the printed circuit board developed for the USB audio codec is shown in **Figure 3**. The board is available ready-made through the Publisher's Readers Services. All connections on the board have been kept as short as possible to keep the unit as compact as possible. As you have come to expect from a high-quality Elektor board, all connectors are fitted at the edge(s).

The vast majority of components that make up the circuit are off the shelf types and their mounting is unlikely to present problems. However, working with SMDs requires some getting used to, and that is why we recommend starting with these parts. With IC2, the pin leading is such that soldering is not too difficult. IC1, however, requires a steady hand and a solder iron with a fine tip. Carefully position the IC on the board and first solder diagonally opposed pins. Then check if the IC body needs re-positioning. If not, solder the other pins quickly and not using too much solder. Use fresh desoldering braid to remove excess solder and do not forget to let the IC cool down in between solder actions.

Besides integrated circuits the PCB also contains a couple of SMDs — supply filtering chokes L1 and L2, and four decoupling capacitors for a number of supply internal to the PCM2902. Of these four miniature 1-?F capacitors, C20, C25 and C26 are mounted at the top side of the PCB, in the space between IC1 and quartz crystal X1 (see the close-up photograph in **Figure 4**). Capacitor C19 is mounted at the solder side of the board, between two through-plating spots. Chokes L1 and L2 also end up at the underside of the board (see **Figure 5**). The board allows SMDs with 06003 cases as well as the larger 0805 cases to be fitted. The latter are preferred because they are larger and therefore easier to handle.

Software installation

The installation of the board under Windows 98SE is unlikely to cause problems. First you will be notified that a 'USB Audio Codec' has

COMPONENTS LIST

Resistors:
R1,R3 = 6kΩ49
R2,R4 = 3kΩ92
R5 = 4-way 1kΩ5 SIL array
R6,R7 = 22Ω
R8 = 1MΩ
R9,R16 = 5kΩ76
R10,R17 = 6kΩ81
R11,R18 = 6kΩ65
R12,R19 = 20kΩ0
R13,R20 = 10kΩ0
R14,R21 = 100Ω
R15,R22,R25 = 100kΩ
R23 = 4Ω7
R24 = 8kΩ2
R26 = 33kΩ
R27 = 1kΩ5

Capacitors:
C1,C2,C14,C21 = 3µF3 50V, MKT (Siemens), lead pitch 5 or 7.5mm
C3,C4 = 22pF
C5,C12,C13 = 10µF 63V radial
C6,C23,C24,C28 = 100nF ceramic, lead pitch 5mm
C7,C8 = 33pF
C9,C16 = 1nF5 1% *
C10,C17 = 1nF 1% *
C11,C18 = 470pF 1% *
C15,C22 = 47pF
C19,C20,C25,C26 = 1µF 25V, SMD case 0805 (e.g., Farnell # 317-640)
C27 = 100µF 10V radial

*) polypropylene or polystyrene (EMZ)

Inductors:
L1,L2 = BLM31A601S, SMD case 1206 (e.g., Farnell #. 581-094)
L3,L4 = 47µH

Semiconductors:
D1 = LED, red, high efficiency
T1 = BC557B
IC1 = PCM2902 (Texas Instruments/Burr-Brown)
IC2 = OPA2353UA (Texas Instruments/Burr-Brown)
IC3 = TORX173 (Toshiba)
IC4 = TOTX173 (Toshiba)
IC5 = LE33CZ (ST) (e.g., Farnell # 302-4568)

Miscellaneous:
JP1,JP2 = 3-way SIL pinheader with jumper
JP3,JP4,JP5 = 2-way SIL pinheader
K1,K2 = 3.5mm stereo jack socket, PCB mount (Conrad Electronics # 73 28 93-88)
K3 = USB connector, PCB mount, type B
X1 = 12 MHz quartz crystal, parallel resonance, (Cload = 30pF)
PCB, order code **020178-1** (see Readers Service page)

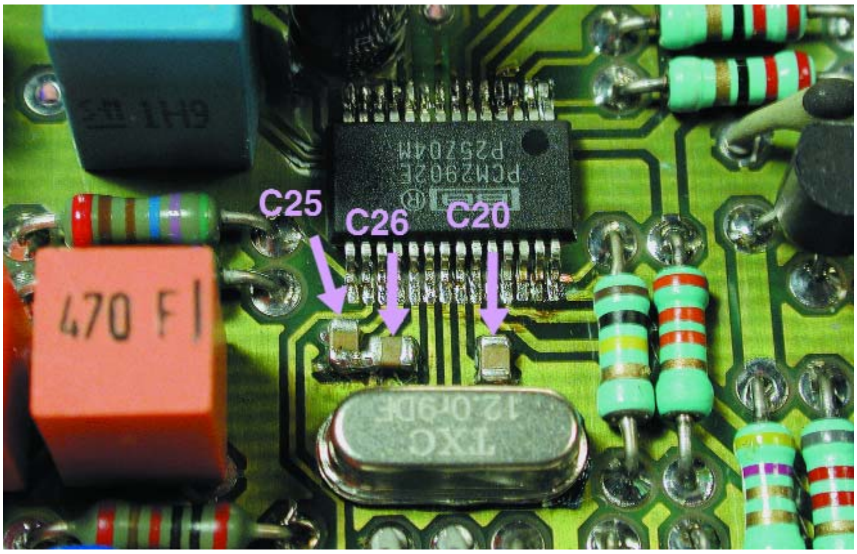


Figure 4. Mounting IC1, C20, C25 and 26 requires precision in soldering.

been found, which requires drivers to be installed. Next, three devices are found (keep the Windows CD-ROM handy): a 'USB Composite Device', a 'USB Human Interface Device' and finally a 'USB Audio Device'.

These three items may be found back under the System Properties/Device Manager tab, see the screendump in **Figure 6**. Next, with Playback properties you should see three slide controls: Desktop, Wave and SW Synth. A fourth slider called 'CD Player (for Window's own simple software player for audio CDs), will be missing and is not found back until the PC is restarted.

In Control Panel, under the Multimedia/CD Music tab, do not forget to check the 'Enable digital CD audio for this CD-ROM device'. When the USB link is briefly interrupted, the control will probably disappear. This is normal (at least, for Windows) and caused by a bug in the operating system. The problem has not been solved under Windows ME or XP, which use different names for some of the windows and controls.

For a quick test of the HID interface, open the volume control win-

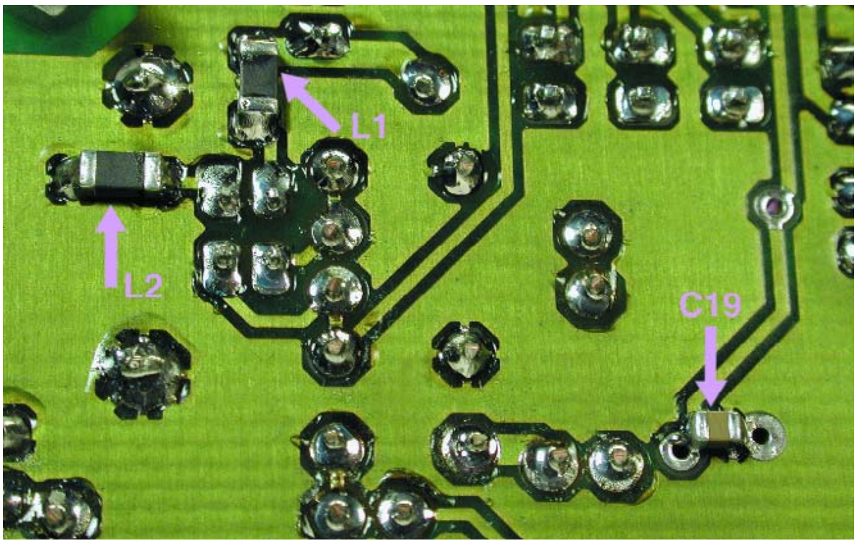


Figure 5. Just like C19, SMD chokes L1 and L2 are fitted at the solder side of the board.

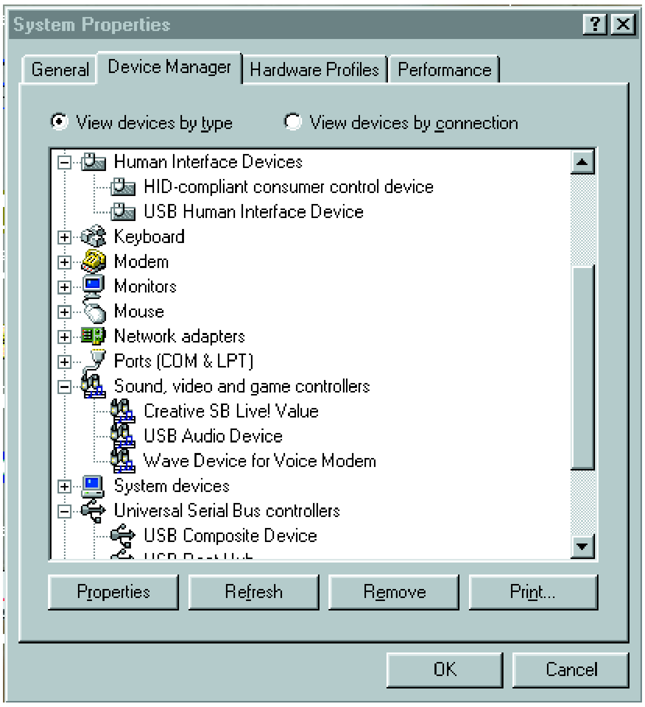


Figure 6. The software installation is quite straightforward (Windows 98SE).

dow. If you establish a contact between the pins of jumpers JP3, JP4 and JP5, you will see 'Mute' going on and off with the loudspeaker slider, while the slider itself moves up or down.

Tip

Those of you who want to test this little sound card may find freeware and shareware on the Internet. During our 'quest' we came across 'audioTester at this website:

www.sumuller.de/audiotester

This software turns your codec into an audio test system comprising a spectrum analyser, an oscilloscope and a signal generator. We tested version 1.4h which comes highly recommended.

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If a picture...

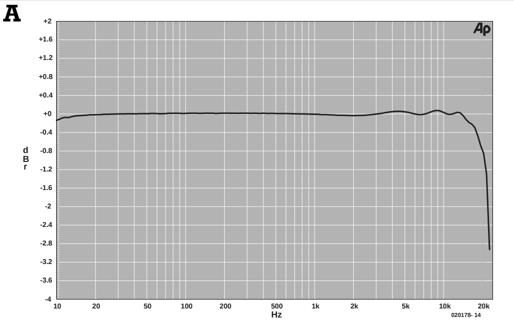
A number of test graphs were plotted that show the equivalent of two pages of text or more.

The first graph (**A**) is practically oriented showing the audio level response of the ADC and the DAC connected in series. To make this measurement we employed the ASIO Multimedia Driver from Cubase VST/32. The standard USB audio drivers in Windows do not allow full-duplex operation, hence prevent you from listing in on the recording! Oddly they do allow an audio CD or a wave file to be played at that time! At 20 kHz, the audio level has dropped by just 0.8 dB. Because the graph scale has been enlarged, the ripple caused by the digital filter of the DC is clearly visible — it is not caused by the analogue filter!

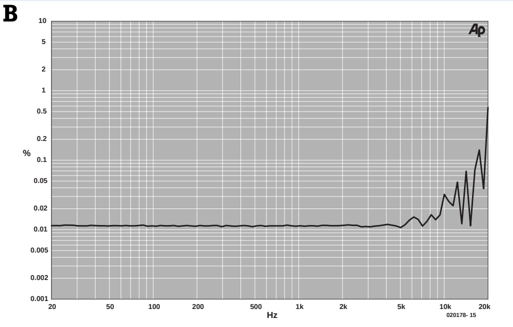
Graph **B** shows distortion and noise as a function of frequency. Here, too, the ADC and DAC are measured in series. The distortion is mainly caused by the ADC, and closer scrutiny using FFT analysis indicates the presence of other mixing products besides the expected harmonic distortion. These products are probably generated by the internal PLL and the digital filters. From 5 kHz onwards, the aliasing products contribute most to the overall distortion. However, they do not have alarming levels given the simplicity of the circuit.

Curve **C**, finally, shows the frequency spectrum of 997 Hz with full drive of the DAC (using a test CD). All harmonics are below –90 dB and account for nearly all of the 0.005% distortion measured. All noise above 20 kHz is typical for delta-sigma converters using noise sampling, and is not worrying if a measurement bandwidth of 22 kHz is used. The analogue output filter limits the larger part of the noise and aliasing products above 35 kHz to acceptable levels. If the bandwidth of the THD+N measurement is extended to 80 kHz, these products will of course cause the overall distortion to increase (to 0.046%).

A



B



C