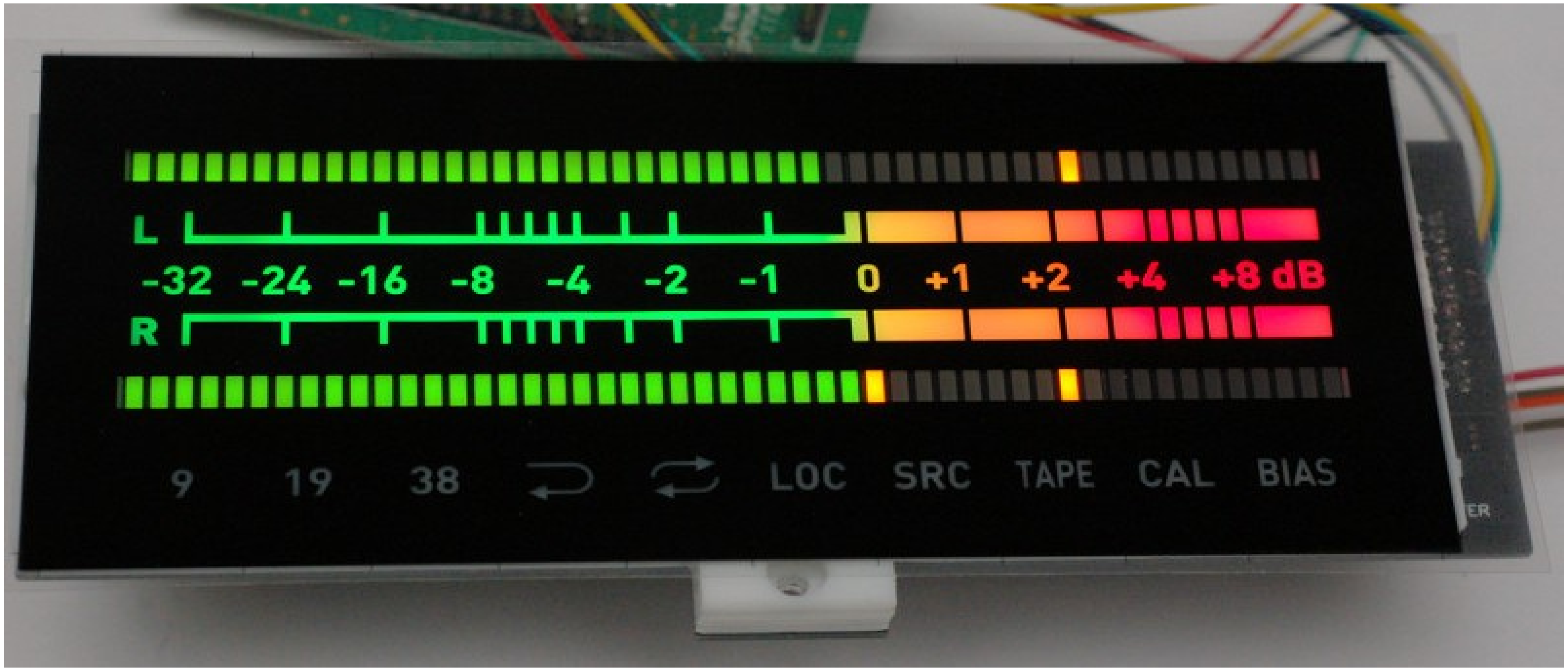
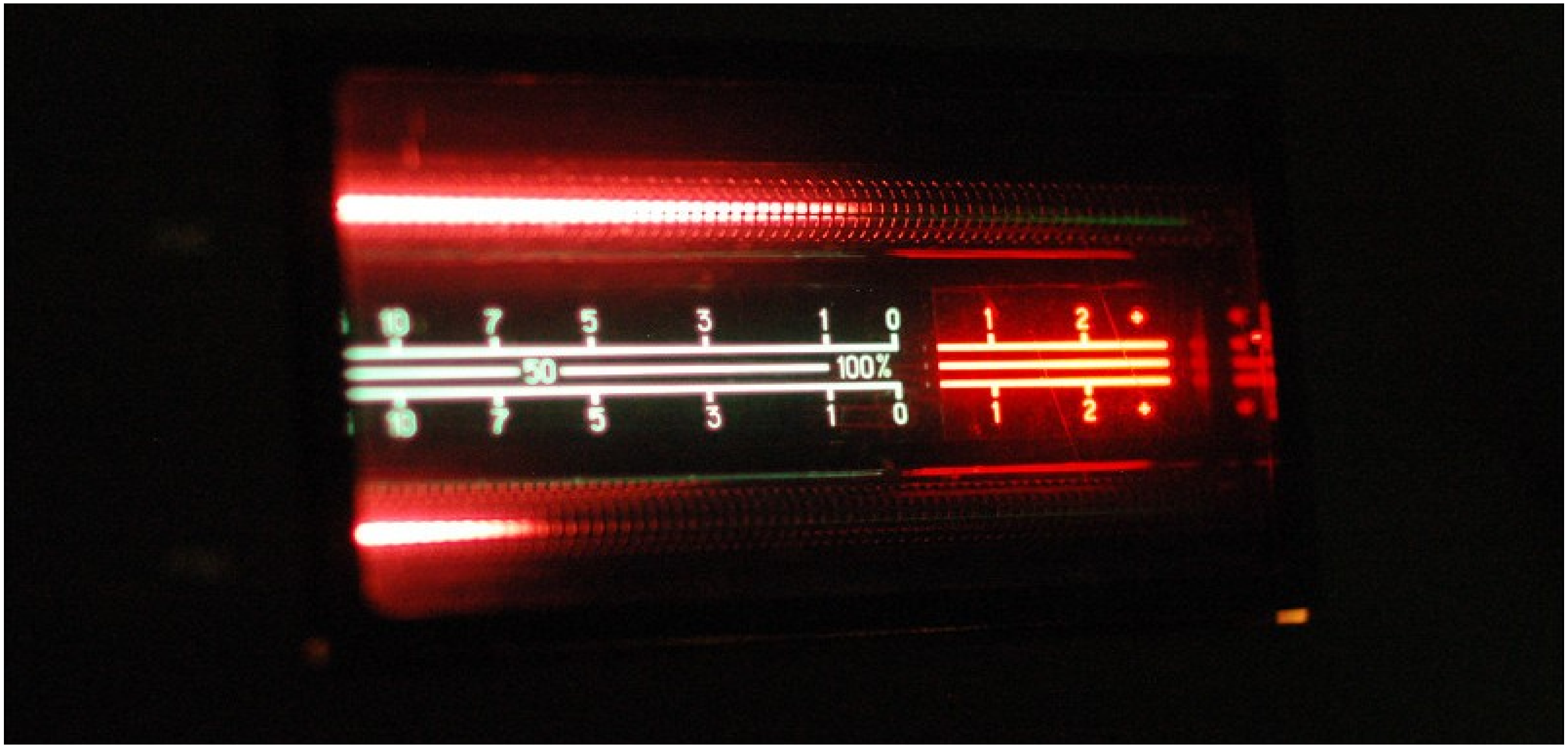
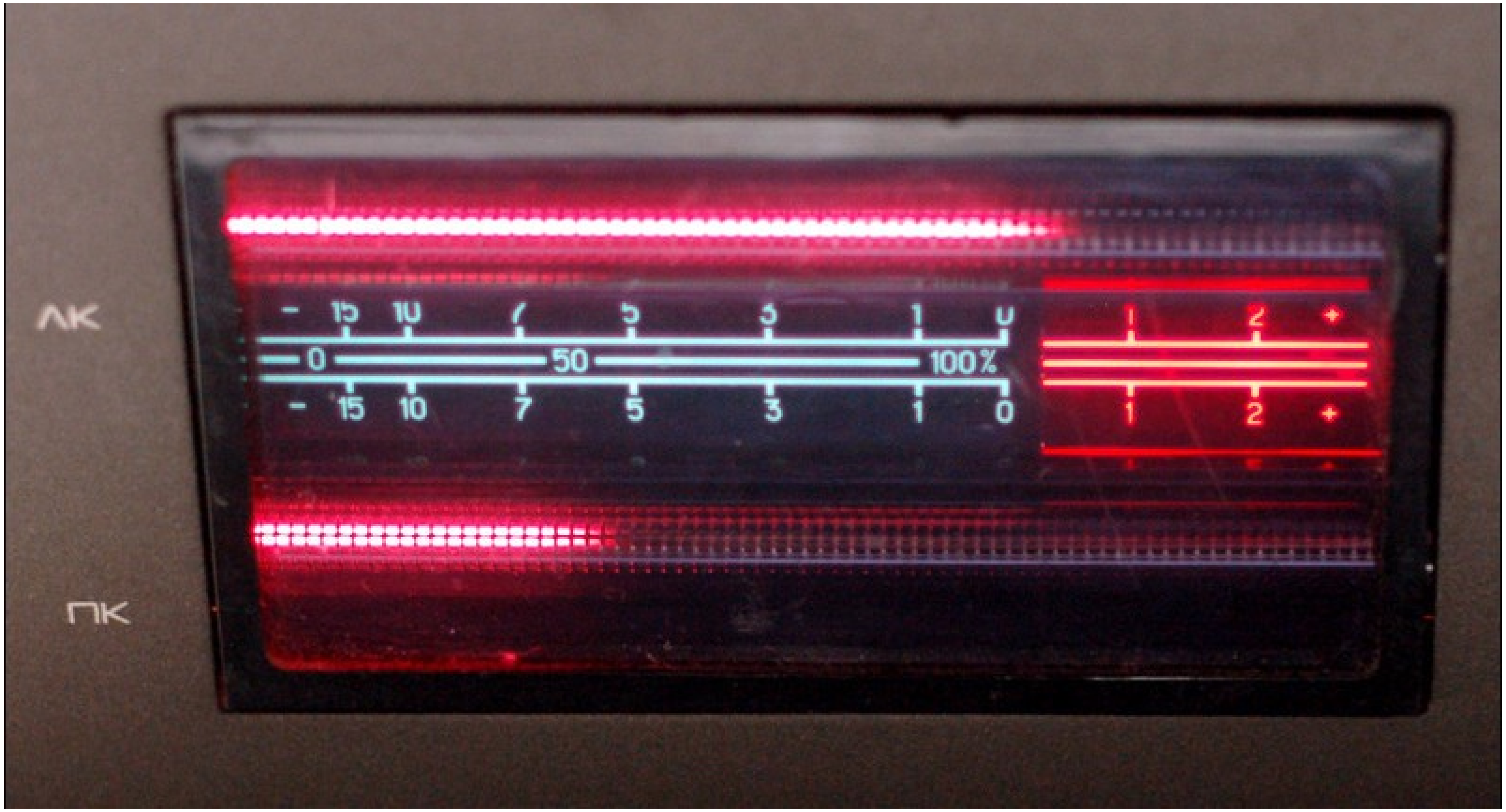
Level meter



The standard meter of the signal level of the "Electronics-004" tape recorder is implemented on gas-discharge linear indicators of the IN-13 type. Such indicators were the hallmark of the early models of the "Electronics" and "Olympus" tape recorders.



For that time it was progressive - bars with continuously varying lengths. But today this dim and blurry indicator does not look at all against the background of the new LED indication of the tape counter.



Indicators IN-13 have a number of disadvantages: low luminosity, fuzzy border of the light column, low accuracy. The disadvantages of level meters based on such indicators are indicated in the literature [1]:

Точность считывания показаний ЛГИ определяется зависимостью длины светящегося столбика L от тока через прибор I, называемой рабочей характеристикой.

Если исходить из постоянства плотности тока на проволочном катоде, то эта характеристика должна иметь вид прямой линии, проходящей через начало координат. Однако реальная рабочая характеристика ЛГИ в начале и конце имеет нелинейные участки (рис. 7.5 *б*). Поэтому для измерения используется средняя часть характеристики, которая может быть представлена в виде сдвинутой по отношению к началу координат прямой линии. Кроме того, в реальных ЛГИ длина свечения L отличается от теоретического значения $L_{теор}$, определяемого по рабочей характеристике. Разность $\delta L = L - L_{теор}$ называется абсолютной погрешностью рабочей характеристики. Величина δL в основном связана с микронеоднородностями эмиссионных свойств поверхности катода, влияющими на параметры тлеющего разряда. Для обеспечения линейности рабочей характеристики необходимо при изготовлении ЛГИ обеспечить постоянство диаметра, чистоту материала и газового наполнения, однородность структуры катода и тщательную технологическую обработку поверхности катода.

На точность ЛГИ оказывает также влияние температура окружающей среды. С ростом температуры уменьшается наклон рабочей характеристики. С целью снижения влияния температуры в качестве основного компонента газового наполнения ЛГИ используется гелий как инертный газ с высокой теплопроводностью.

С учетом изложенных факторов практически погрешность показаний ЛГИ оказывается не менее 5—10%.

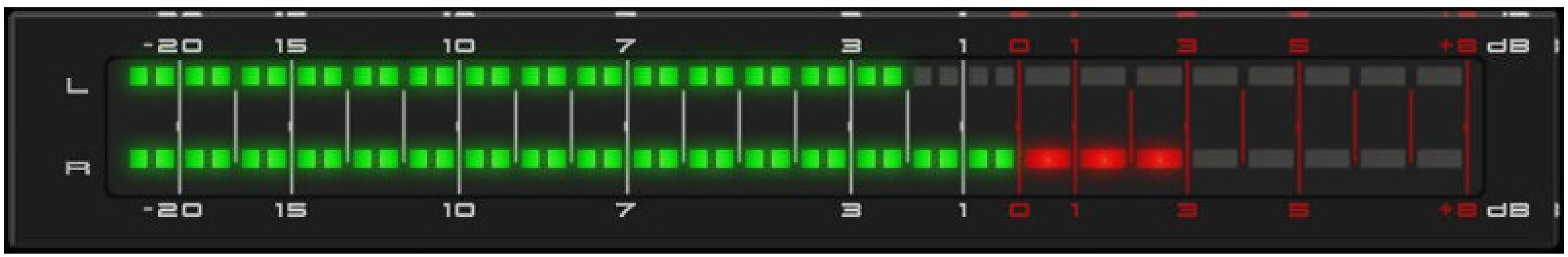
Газоразрядные трубки, в том числе получившие широкое распространение трубки типа ИН-13, имеют недостаточную яркость, нечеткую границу светового столбика, длина шкалы (110 мм) также маловата.

There are different options for display elements. These are dial gauges, gas-discharge indicators, vacuum-luminescent indicators, LED and liquid crystal indicators. There are also more exotic options, for example, "bunny" devices [1]:

Какой же из типов показывающих приборов является наиболее предпочтительным? Это определяется прежде всего местом установки ИУ. Так, в результате опроса звукорежиссеров ГДРЗ и ТТЦ было выяснено, что оптимальными для их работы являются «зайчиковые» ИУ. Они имеют оптимальные размеры (примерно 160 мм), достаточную яркость, высокую четкость «зайчика» (светового штриха). Стрелочные приборы менее удобны — они меньше по размерам, практически непригодны для контроля уровней стереофонических передач, по ним труднее производить отсчет.

In addition to the drawbacks of the IN-13 indicators themselves, the standard level meter has other drawbacks. To illuminate the scale, incandescent bulbs are used, which have a low brightness and a limited service life. The control scheme also has a number of disadvantages. Half-wave rectification of the audio signal is applied, which can lead to errors in level measurement. The integration time constant does not correspond to the value accepted by the standard, which manifests itself in the form of an underestimation in comparison with the reference devices. The range of indicated signal levels is not wide enough, especially taking into account the overload capacity of modern tapes.

At the end of the 1980s, microcontrollers were already in full swing to control the devices, and along with electronic counters on LED indicators, LED level meters were also used. For example, they were used by the flagship Akai GX-747. By the way, the appearance of this meter is quite deceiving - each green LED is divided into two parts and it seems that these are two separate LEDs. In fact, there are only 24 of them per channel.



It is most convenient for a level meter to use ready-made LED assemblies. The most common are 10-piece assemblies, but there are also 8, 5, or 4-piece assemblies.

Unlike the level meter on IN-13, which has no column resolution, LED meters can change the column length only discretely. To minimize this drawback, the number of items in the rulers should be large. If you use LED assemblies with a pitch of 2.54 mm, then a line of 35 elements is placed in the standard indicator window. Pretty good, but I would like more.

If you look closely at the design of the front panel of the "Electronics-004" tape recorder, you will notice a large amount of empty space around the level meter.


ЭЛЕКТРОНИКА-004
-СТЕРЕО
ЭЛЕКТРОННОЕ УПРАВЛЕНИЕ
4 МАГНИТНЫЕ ГОЛОВКИ
РЕВЕРС
0000
СДЕЛАНО В СССР
УРОВЕНЬ ЗАПИСИ
ГРОМКОСТЬ

This drawback can be corrected, and with an advantage - the rulers of the meter can be made longer. To install a new level meter, you will need to enlarge the window in the front panel. A similar operation was required to install an electronic tape counter. Another window is required for the remote control photodetector. All these windows are covered with dark plexiglass. As a result, 3 new rectangular windows appear on the panel, located almost diagonally. These windows do not affect the areas of the panel where there are labels. In general, the appearance of the tape recorder becomes better while maintaining the same style.


ЭЛЕКТРОНИКА-004
-СТЕРЕО
ЭЛЕКТРОННОЕ УПРАВЛЕНИЕ
4 МАГНИТНЫЕ ГОЛОВКИ
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СДЕЛАНО В СССР
УРОВЕНЬ ЗАПИСИ
ГРОМКОСТЬ

In a new window, you can place a caliper with a ruler of 50 elements. This will make it possible to almost completely get rid of the effect of discreteness in the change in the length of the luminous column. Its movements will be as smooth as for the continuous bars of the gas-discharge indicator. At the same time, a number of advantages appear: the brightness is significantly higher, the border of the column is clear, the column itself is multicolored, it is possible to realize the memorization of peaks, and much more. A scale with multicolor illumination is placed between the rulers, and at the bottom there are additional stencils that allow you to display various modes of operation of the tape recorder.


ЭЛЕКТРОНИКА-004
-СТЕРЕО
ЭЛЕКТРОННОЕ УПРАВЛЕНИЕ
4 МАГНИТНЫЕ ГОЛОВКИ
РЕВЕРС
-9.59.59.9
СДЕЛАНО В СССР
УРОВЕНЬ ЗАПИСИ
ГРОМКОСТЬ
L
R
0
+1
+2
TAPE
CAL
BIAS

The main characteristics of level meters are integration time and return time. The integration time determines how quickly the meter responds to rapid changes in signal level. If the integration time is long (about 300 ms), you get an average level meter: the so-called VU-meter. Such a meter will not react to short signal peaks, which can cause congestion of the recording-playback channel. Therefore, its use is undesirable. When dial gauges were used as an indicating device, the integration time could not be made small due to the inertia of the moving system. Therefore, the VU-meter was often combined with a faster peak threshold indicator on one or more LEDs. Application of high-speed display devices, such as gas-discharge indicators or LEDs,removed the problem of obtaining short integration times and made it possible to build peak or quasi-peak meters.

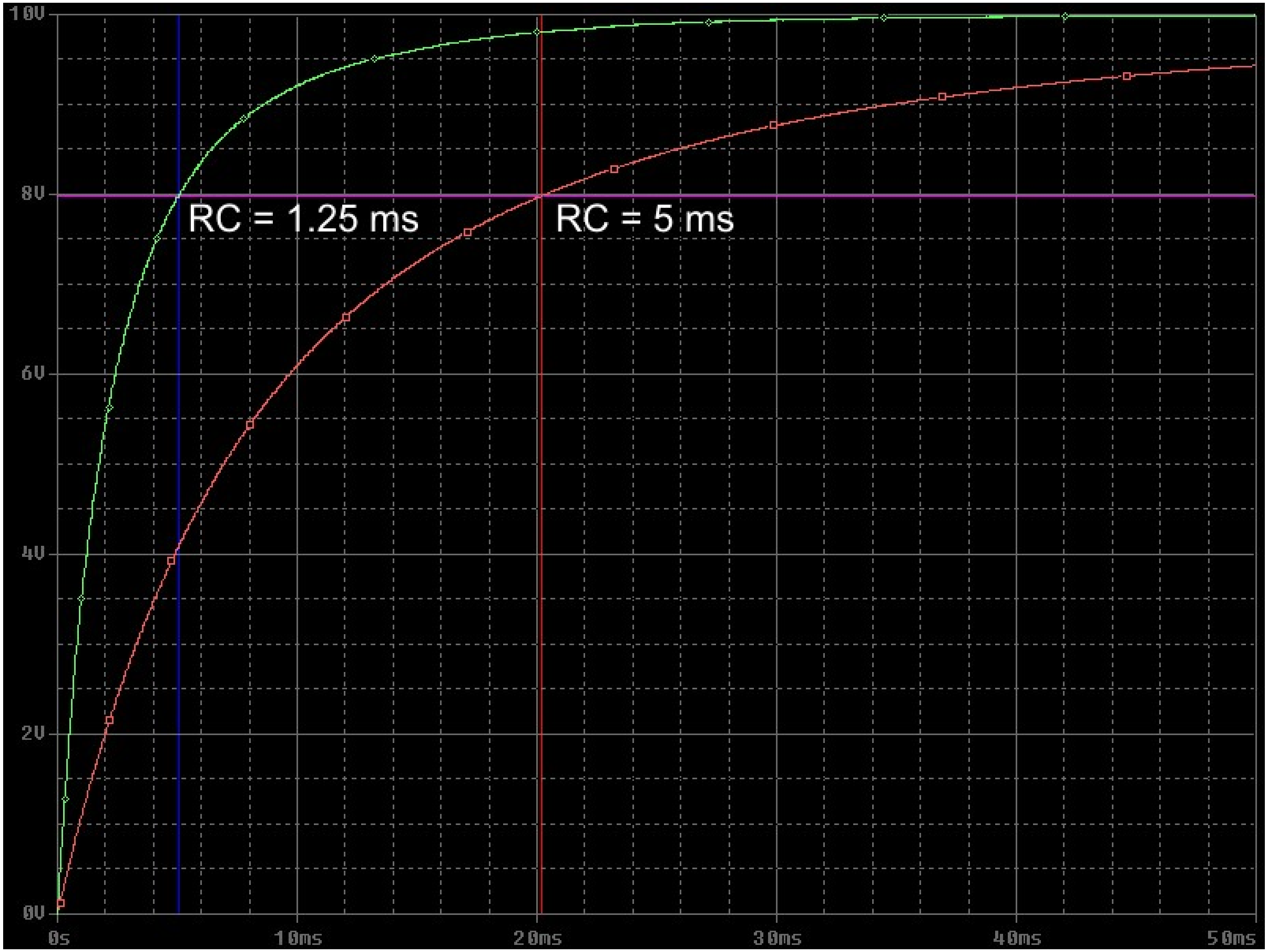
Most often, quasi-peak level meters are used to measure the signal level in audio paths. Unlike true-peak meters, which in theory have zero integration time, for quasi-peak meters this time is specified by standards.

It would seem that it is necessary to strive for the shortest possible integration time so that the indicator can register the shortest signal peaks, avoiding overloading the path. This approach is used in digital paths, where even a short-term overload leads to undesirable consequences. Therefore, the indicators of the peak level are usually used there. For analog magnetic recording, the momentary overload may not be heard at all. On the contrary, if you strive to completely get rid of the overload at the signal peaks, you will have to underestimate the average recording level, which will lead to a more noticeable problem by ear - a deterioration in the signal-to-noise ratio ... Therefore, for analog magnetic recording, it makes sense to choose some optimal value for the integration time of the level meter,so that it allows short-term overloads, but only those that are invisible to the ear.

Integration time for quasi-peak meters is defined as the length of a single 5 kHz tone burst at which the reading reaches -2 dB (approximately 0.8) of steady state. This definition is given by GOST 21185-75 and IEC 60268-10: "... the duration of a burst of sinusoidal voltage of 5000 Hz at reference level which results in an indication 2 dB below reference indication". These standards define an integration time of 5 ms for quasi-peak meters. The earlier GOST defined the integration time of 10 ms, but at the same time the level of -1 dB had to be reached, which practically corresponds to the values of 5 ms and -2 dB, ie there is no difference in these standards.

Quasi-peak meters are usually constructed as follows: the input signal is fed to a rectifier, preferably a full-wave rectifier, at the output of which a signal equal to the modulus of the input signal is obtained. This is followed by a peak detector and an RC smoothing circuit with different charge and discharge time constants. The charging time determines the integration time, and the discharge time determines the return time. The mid-level meters did not have a peak detector, where the signal from the rectifier was fed directly to the smoothing RC circuit. Therefore the VU-meter has only one time constant.

The integration time is not numerically equal to the charging time constant of the RC smoothing circuit. If at the output of the peak detector you turn on the RC-circuit with a charging time constant of 5 ms, then the level of 0.8 will be reached in about 20 ms. Simulations show that 0.8 is reached in 5ms for a chain with a charge time constant of about 1.25ms. Those. the integration time is approximately 4 tau of charging the RC circuit.



That, however, can be read in the books [1].

Время интеграции для всех ИУ задано одинаковым — 5 мс.

Как видно из табл. 5.1, при степени заряженности конденсатора 0,8 коэффициент, характеризующий соотношение $t_з/\tau_з$, равен 4,02. Следовательно, постоянная цепи заряда квазипиковых ИУ равна $\tau_з = t_з/4{,}02 = 5 \cdot 10^{-3}/4{,}02 = 1{,}25$ мс.

Nevertheless, in the standard circuit of the "Elektronika-004" tape recorder, the charging time constant of the RC-chain is close to 5 ms. This already gives a 4 times longer integration time compared to the standard one. But the situation there is even worse - a half-wave rectifier is used, as a result, the level of 0.8 is reached in about 40 ms! This is probably the reason for the noticeable difference in readings on the real music signal of a standard meter and an external meter such as RTW 1206N, where the integration time is kept exactly. Although when calibrated on a sinusoidal signal, their readings are the same.

In addition to the integration time, quasi-peak meters also have a return time, which is much longer. This time determines how quickly the reading will decrease after the signal is removed. If this time is made small (for example, equal to the integration time), then the indicator readings will "jerk" too quickly, making them difficult to read. For mid-level meters, such a problem did not arise due to their low speed. One time constant could be dispensed with. In high-speed quasi-peak meters, it is necessary to artificially slow down the reset of readings so that the operator can read the information.

The return time is defined as the time after which the indicator reading decreases by 20 dB after the signal is removed. It is also not equal to the discharge time constant of the RC circuit, but is approximately 2.3 tau. The return time is also specified by the standards. It differs for indicators of different purposes. For indicators of the first type, which serve to control the signal level during its operational adjustment (this is just the case of adjusting the recording level in a tape recorder), the return time should be 1.7 ± 0.3 sec. Accordingly, the time constant of the discharge of the RC-chain should be approximately 740 ms.

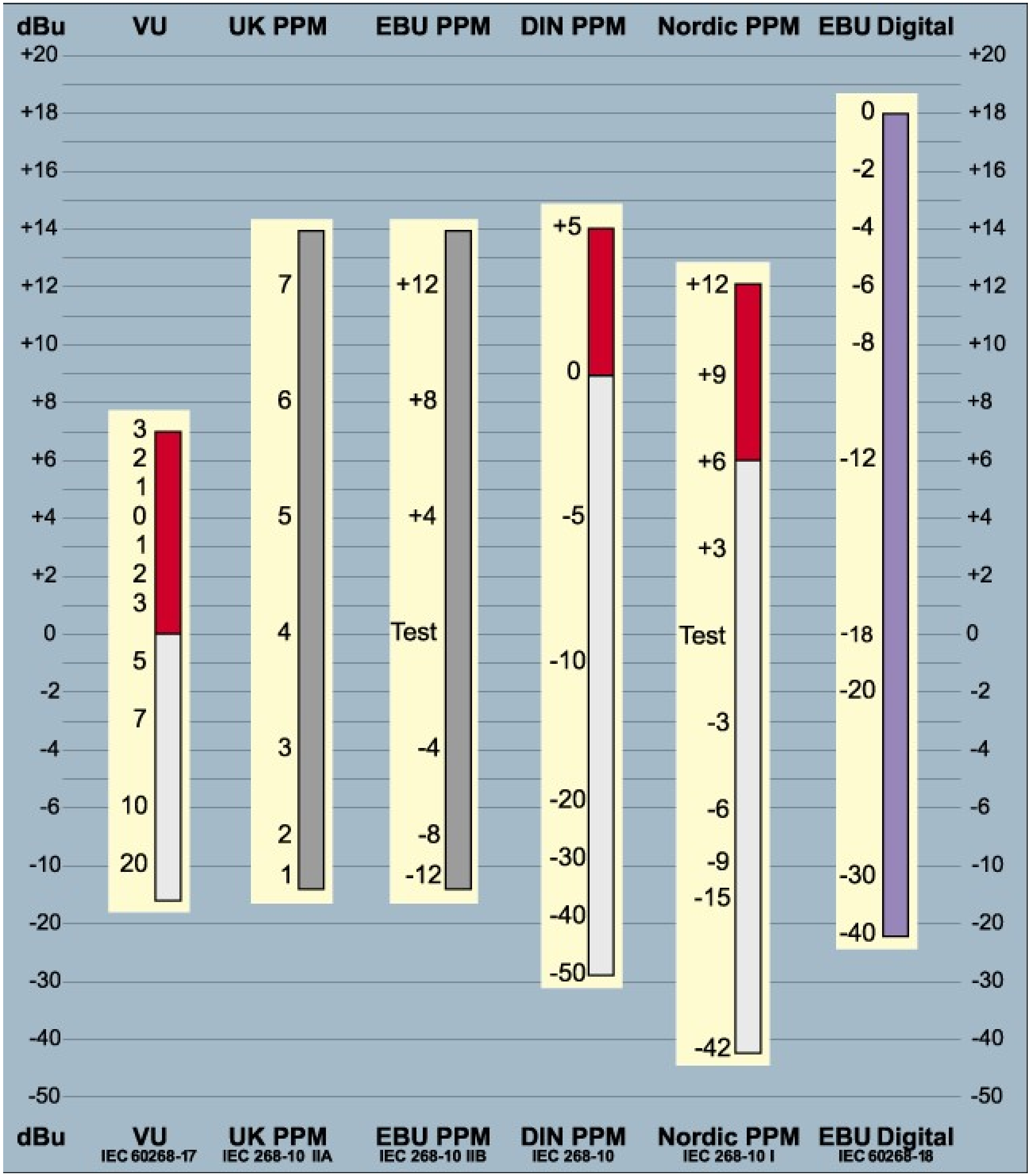
Audio level meters are often built on the basis of analog circuits. The design turns out to be rather cumbersome and contains nodes that are rather difficult to implement with good accuracy: a peak detector, a linear-logarithmic converter. Sometimes some of the functions are transferred to the microcontroller, which most often digitizes the ready-made DC voltage after the detector. This method is easy to implement, but it has a number of significant disadvantages. Since the detector is hardware-based, it is impossible to change its type and characteristics. There is only one detector, which does not allow making a combined meter (for example, average + peak level). For detecting small signals, a diode detector is not suitable due to poor linearity, you have to make an active detector on an op-amp. In this case,the bias voltage of the op-amp limits the dynamic range of the meter, and nothing can be done about it.

Recently, it has become possible to carry out all signal processing inside the microcontroller, feeding the audio signal directly to the ADC input. Common AVR microcontrollers are not very well suited for this task, as they have a relatively slow ADC (maximum 15 kHz at 10 bits). More suitable are STM32 controllers.

In digital implementation, the detection process does not introduce DC voltage errors. Therefore, it is sufficient to remove the DC component of the AC input signal. The type of detector and the values of the time constants can be easily changed, which makes it possible to implement several modes of operation. For example, VU, RMS, QPPM, True Peak. It is possible to realize the simultaneous display of the average values in a continuous column, and the peak values - in a single burning segment. For each display variant, the integration and retraction times, as well as the display form (column, point, point reversal or disappearance, etc.) can be set independently. Non-volatile memory can be used to store the settings. The readings can be converted to a logarithmic scale according to the table,which allows you to create a scale of any kind (for example, S-shaped with an interval stretching near 0 dB).

Existing level meters can be examined to select suitable options for implementation in the meter. Modern programs for audio mastering contain software implementations of various versions of the level meter:

Many of these options (eg Nordic, BBC, EBU) differ only in the choice of the reference level. There may still be differences in the integration and return times, but basically all these meters are quasi-peak. There are quite a few ways to bind the scale by level:


dBu
VU
UK PPM
EBU PPM
DIN PPM
Nordic PPM
EBU Digital
+20
+18
+16
+14
+12
+10
+8
+6
+4
+2
0
-2
-4
-6
-10
-20
-30
-40
-50
3
2
1
0
1
2
3
5
7
10
20
7
6
5
4
3
2
1
+12
+8
+4
Test
-4
-8
-12
+5
0
-5
-10
-20
-30
-40
-50
+12
+9
+6
+3
Test
-3
-6
-9
-15
-42
0
-2
-4
-6
-8
-12
-18
-20
-30
-40
IEC 60268-17
IEC 268-10 IIA
IEC 268-10 IIB
IEC 268-10
IEC 268-10 I
IEC 60268-18

With reference to an analog tape recorder, the zero level of the indicator is tied to a certain magnetization of the tape, which is taken as nominal. For household reel-to-reel tape recorders operating at a speed of 19.05 cm / s, it is usually equal to 320 nWb / m.

Ideally, for each specific tape, at the calibration stage, the maximum recording level should be determined according to the criterion of a given level of nonlinear distortion. In proportion to this level, the zero of the indicator should also be adjusted. Then, for any tape, you can use the entire available dynamic range. But such a complex calibration procedure has been implemented only in rare devices, for example, in the cassette BeoCord 9000 from Bang and Olufsen. With a digital level meter, such a calibration is easier, the scale shift can be done by simply multiplying the digital values by a given factor.

As mentioned above, a quasi-peak level meter is best suited for an analog tape recorder. Alternatively, a combination of a VU meter VU and a quasi-peak, or two quasi-peak with different integration times is possible. RMS meters are more suitable for assessing the loudness level and do not make much sense for magnetic recording tasks. In the modern standard ITU-R BS.1770, frequency weighting is also used to measure the loudness level (LUFS); this mode of operation of the meter can be useful when used as a separate device, and not as part of a tape recorder.

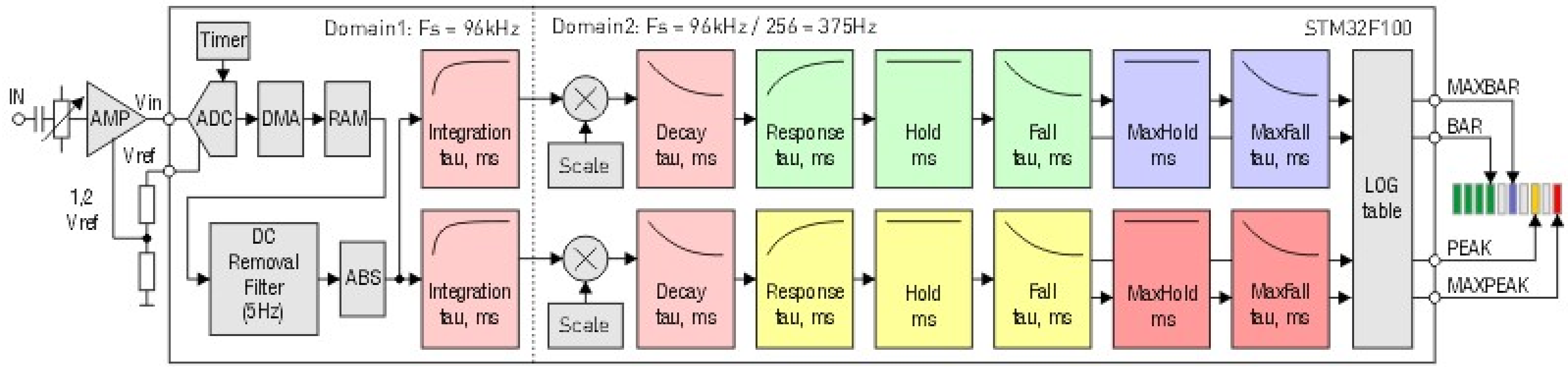
In a tape recorder, frequency weighting is also sometimes used, when in the recording mode the meter displays the level taking into account the frequency correction of the recording current. Then you can see how much headroom is available for a specific input signal with a specific spectral content. For example, this is how the Tandberg TD-20A tape recorder level meter works, it bears the inscription "Equalized peak reading":

Any of these options can be implemented in a digital level meter, the only question is choosing a processor with suitable processing power.

Meter structure

For a level meter with digital signal processing, the analog part is extremely simplified. The microcontroller's ADC input is supplied with an alternating sound signal shifted by about 1/2 Vref. Using DMA codes, the ADC is used to fill the buffer in RAM. When the DMA cycle is half-ready, the data is read and processed. The first thing to do with the signal is to remove the DC component using a special filter. Further, full-wave signal rectification is performed, which is reduced to calculating the absolute value. RMS, average level, quasi-peak level, peak level calculators can then follow. For each of the options (except for true peak), the integration time is set using a 1st order IIR filter. All this is done in the ADC sampling rate domain. Since the return time is significantly longer than the integration time,backtracking processing, as well as outputting the measurement result, can be done in a "slower" domain, for example, with 256 times downsampling.

The block diagram shows one stereo channel of the meter. It has two processing branches that work in parallel and can be used to measure the level in different ways, for example, average and peak.

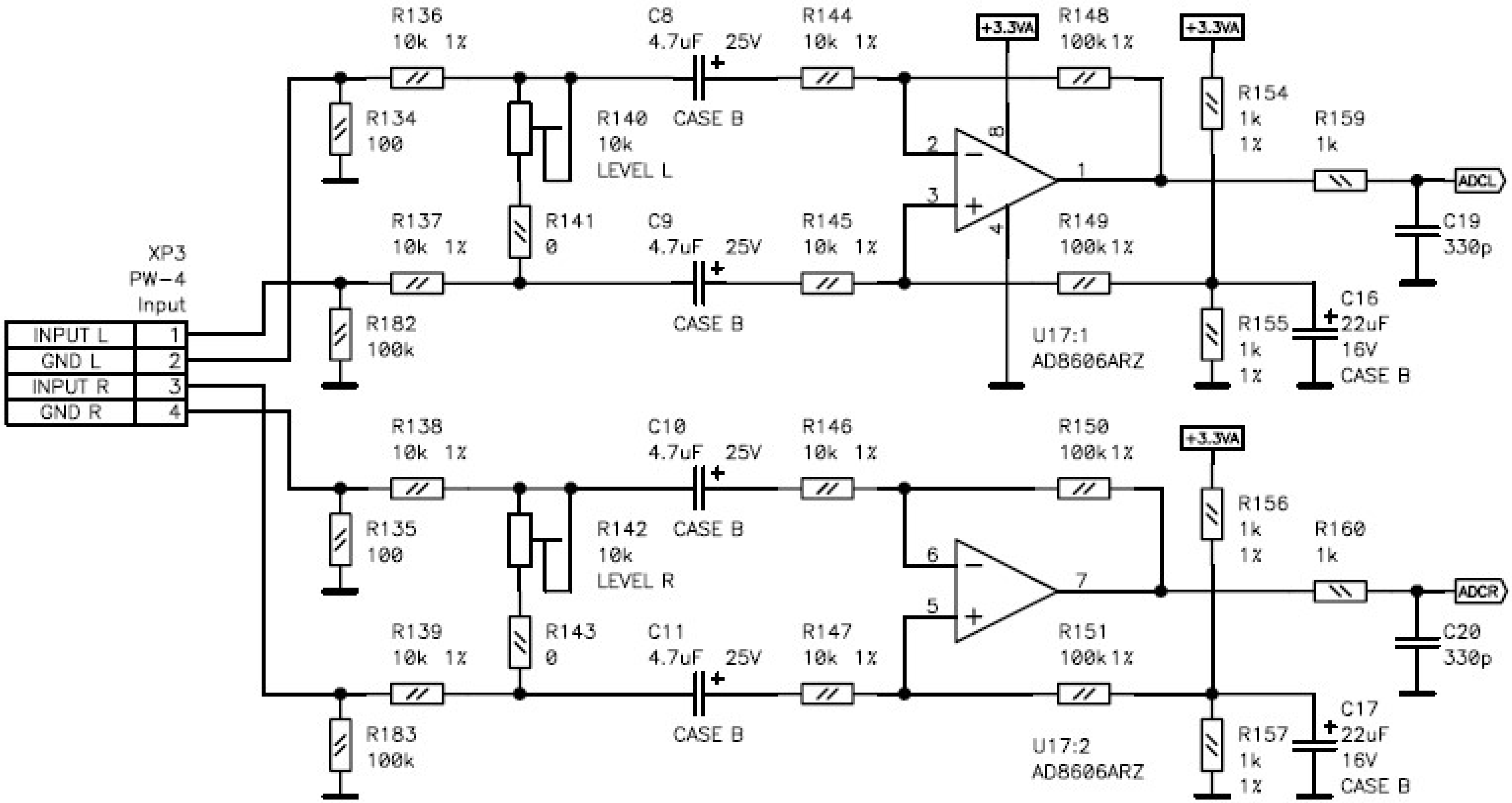


To avoid the need to apply the Anti-alias filter at the ADC input, as well as to be able to register short signal peaks, the sampling rate in the first domain should be chosen rather high. In digital paths, the Inter-Sample Peaks meter skipping problem is solved by at least quadrupling the sampling rate. For an analog tape recorder with a bandwidth of 20 kHz, this corresponds to a sampling rate of about 160 kHz, which is unlikely to have enough resources. Since recording very short peaks is not necessary here, some trade-off sampling rate can be chosen, say 96 kHz.

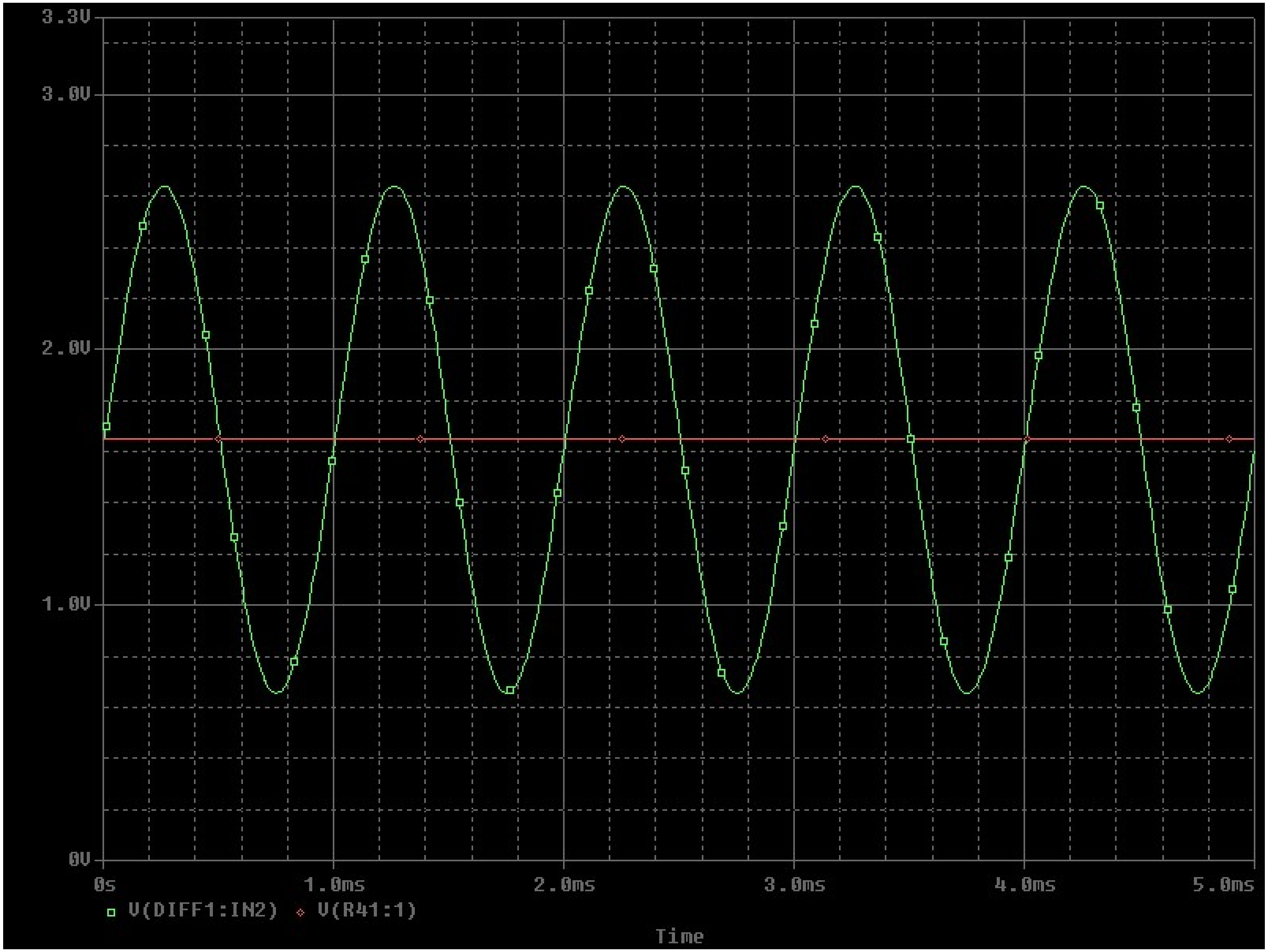
In the "slow" domain, everything is simpler, there, with the help of one more IIR-filter of the 1st order, the return time is formed, which can be set arbitrarily. Then the values are tabularly logarithmized, converted into a positional code, which is loaded into registers. The signal peaks hold branch works in parallel. All this must be done for each stereo channel.

Schematic diagram of the meter

The analog part of the meter contains only a buffer amplifier on an op-amp, which is connected according to the differential amplifier circuit. He immediately solves two problems. First, it allows the output to be shifted by half the ADC scale. Secondly, it allows you to take the input signal differentially.

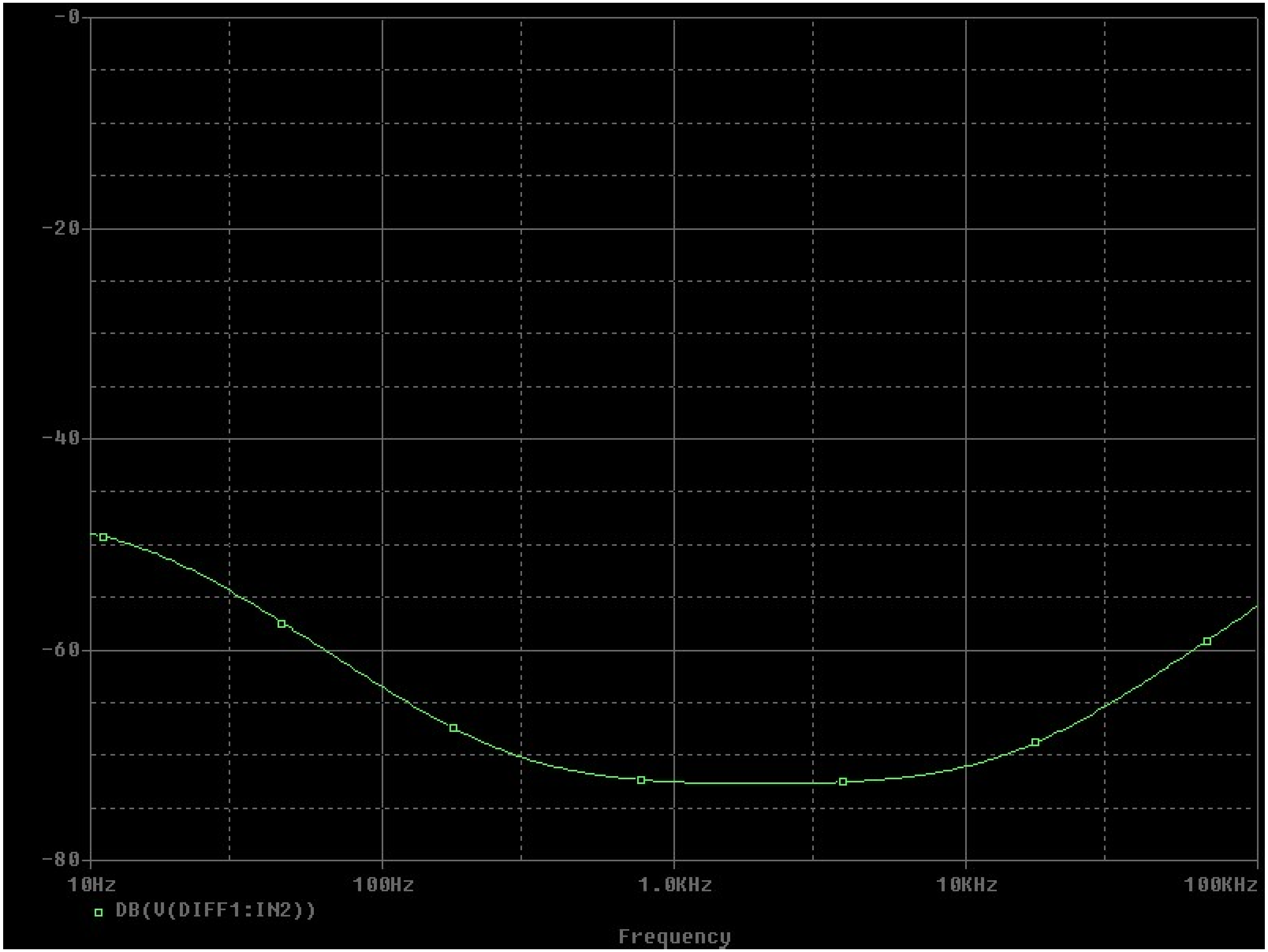


The signal shift is necessary for the reason that the ADC built into the microcontroller is unipolar, its input range is from 0 to 3.3 V (up to the supply voltage VDDA, which is also a reference voltage). The shift is carried out by half of this voltage, i.e. by 1.65 V, which is set by the dividers R154R155 and R156R157. Separate dividers are used to prevent communication between channels, since a resistor divider is not an ideal voltage source. The graph below shows the operation of the input amplifier. The green graph is the op amp output, the red graph is the offset voltage generated by the resistor divider.

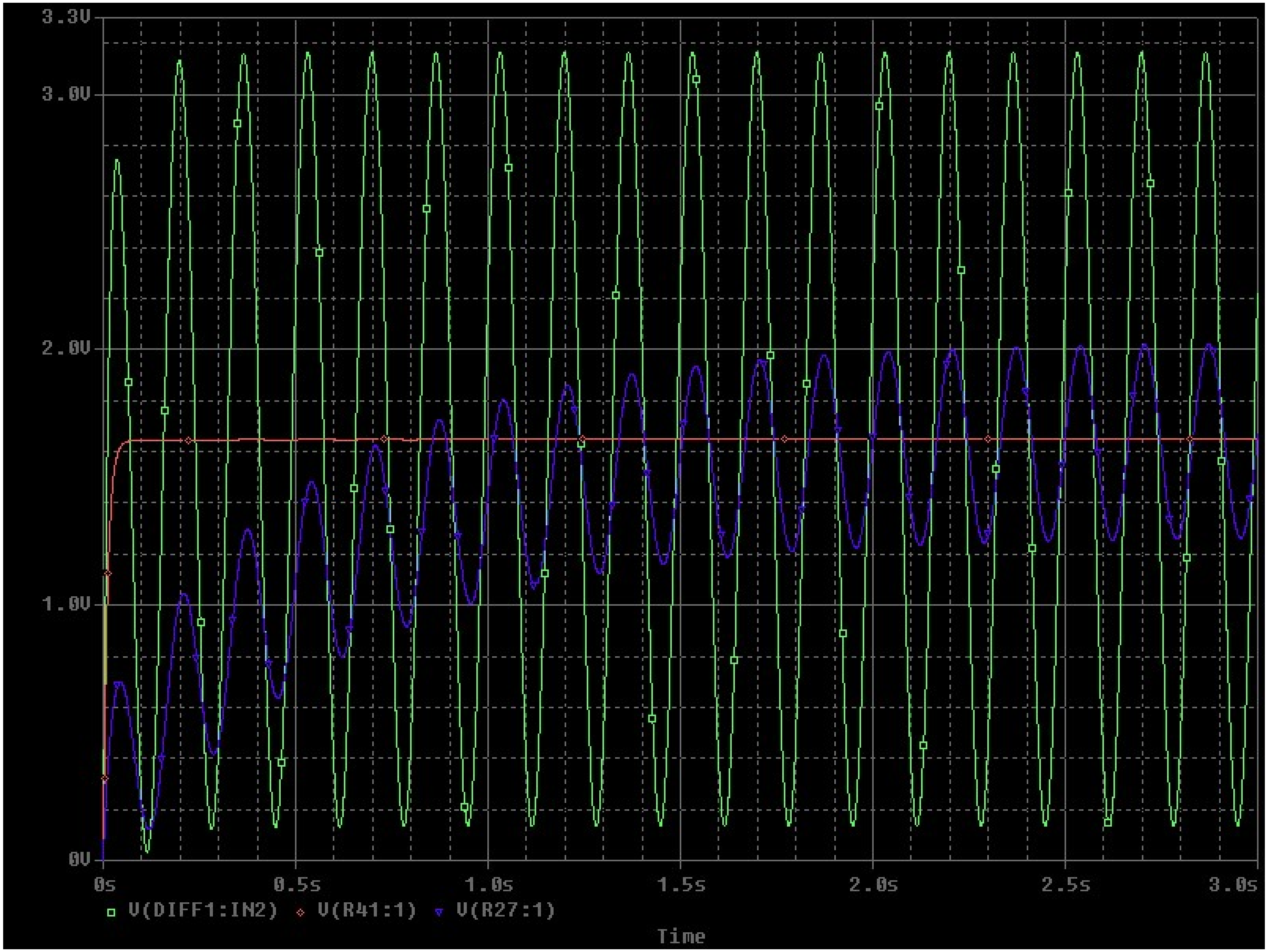


Differential pickup of the input signal is also highly desirable. The level meter is basically a digital device and is powered by a "digital" power supply of +5 V. The current consumption can be quite significant (this is the total supply current of all LEDs), and it can change during operation. As a result, there will be a noticeable voltage drop on the power supply wires, including the ground wire. The ground potential of the level meter can jump relative to the analog ground of the tape recorder. If using a conventional input amplifier, the drop across the earth conductor will be applied to its input. There is no such problem with a differential amplifier. The signal source here is single ended, not differential, so the second input of the diffuser must be connected to the analog ground of the source.the influence of fluctuations in the ground potential of the meter can be eliminated.

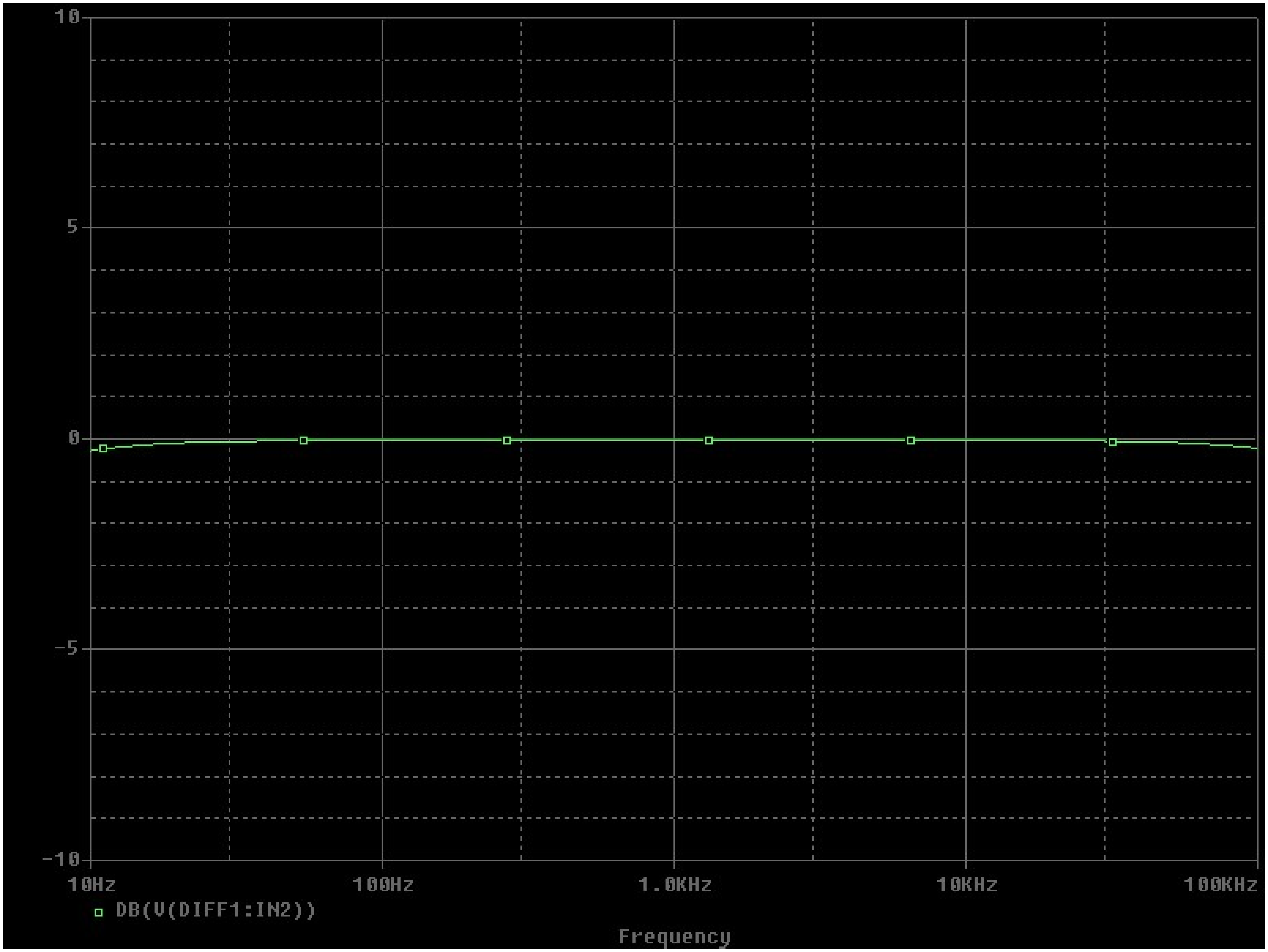
If the level meter is made as a separate device, then you can implement balanced inputs with XLR connectors, as is customary in professional audio equipment. In the graph below, you can see that in the audio frequency range, the common-mode noise rejection of the input amplifier is better than 50 dB, although in reality it will be slightly worse due to the spread of the resistor values. From this point of view, it would be better to use an integral diffuser, for example, of the AD627 type. But it is rarer and more expensive than a conventional op amp. Moreover, in real operating conditions, the suppression of common-mode noise will be sufficient anyway.



This circuit has a not-so-nice feature. The separating capacitors are charged here to approximately the voltage of the divider (1.65 V) during operation. But their charging current passes through rather high-resistance feedback resistors (100 kOhm), although for a useful signal these capacitors form a filter together with lower-resistance resistors (10 kOhm). Therefore, the initial charging of the containers will take much longer than the filter time constant and may take several seconds. In this case, this is not so scary, because in fact the amplifier begins to work normally after 0.2 - 0.3 seconds. For normal operation, it is enough that the voltage at the inputs of the op-amp falls within the permissible range. But for some time after that, the voltage at the inputs of the op-amp continues to be set. For some applications,this may be critical, then the separating tanks will have to be switched on differently. The graph below shows the power-on process of the amplifier. The red graph is the offset voltage generated by the resistor divider. The blue graph is the voltage at the inputs of the op-amp. Green graph - voltage at the op-amp output. It can be seen that the output signal returns to normal much earlier than the process of establishing a constant offset at the inputs of the op-amp ends.It can be seen that the output signal returns to normal much earlier than the process of establishing a constant offset at the inputs of the op-amp ends.It can be seen that the output signal returns to normal much earlier than the process of establishing a constant offset at the inputs of the op-amp ends.



Alternatively, you can simply reduce the capacitance of the blocking capacitors C8 - C11, since the cutoff frequency turned out to be too low: at a frequency of 20 Hz, the attenuation is only 0.1 dB. It is undesirable to use ceramic capacitors here because of the microphone effect; it is better to use tantalum electrolytes.



The complete diagram of the level meter can be found in the file meter50_sch.pdf . The supply voltage of the analog part of the microcontroller (VDDA) is used as the reference voltage for the ADC. This voltage is formed using the U16 stabilizer of the LP2951A-3.3 type. The decoupling of the analog and digital power supply of the microcontroller is carried out by the choke L1 together with blocking capacitors. The analog supply voltage is taken directly from the output of the stabilizer, the digital one - after the choke (according to the recommendations of Analog Devices for mixed-signal IC).

The EEPROM U18 of the 24C16 type is used to store tables and settings. You can switch presets using the JP1 and JP2 jumpers. The presets are loaded using the service program through the XP1 connector of the UART port. Any USB-TTL adapter can be used to connect to a computer. Through the same port, it is possible to download the microcontroller firmware if you turn on the power supply with the JP3 (BOOT) jumper installed. The microcontroller program can be debugged through the XP5 connector, where the signals of the SWD debug interface are output.

LED bars and individual LEDs are connected to the outputs of the shift registers U1 - U14, which are daisy-chained and connected to the microcontroller's SPI port. The indication is static to avoid interference. One of the register output signals is used to control the VT1 key, which allows you to turn off the scale backlight LEDs.

In addition to LED strips, the meter contains additional stencils, which are illuminated by LEDs. They can be used to indicate different modes of operation of the tape recorder. To control them, communication with the control unit is required. It is carried out via the RS-485 interface through the XP2 connector.

Adjustment is required to equalize the brightness of all display elements (rulers, scales and stencils). It can be carried out in software or using potentiometers R180 and R181. Software adjustment can be done in two ways - using PWM or using the built-in DAC of the microcontroller. If a DAC is used, then instead of trimming resistors R180 and R181, jumpers must be installed. If PWM is used, resistors R180 and R181 are not installed. For PWM smoothing, a 3rd order low-pass filter is used, performed on the op-amp U20. From its outputs, the control signals are fed to the adjustable stabilizers U21 and U22 of the LM1117 type. The output voltage VCC1 is used to power the rulers, and VCC2 is used to power the backlight and stencils.

The XP6 and XP7 connectors can receive the output voltage of the built-in DACs of the microcontroller. These outputs can be used to build a level meter based on arrow indicators. Thanks to digital signal processing, it becomes possible to display the quasi-peak level, albeit with some delay due to the inertia of the mobile system. Sluggishness can also be partially compensated by adding software "forcing". The scale of the dial gauges can have any graduation, the dependence is set in a tabular manner, as for the LED version of the meter.

This is where the hardware part of the level meter ends, everything else happens inside the microcontroller.

ADC and DMA

The first step is to digitize the input signal and place it in a buffer in RAM. This will be handled by a built-in 12-bit analog-to-digital converter (ADC) and direct memory access (DMA) controller. The level meter is intended for a stereo device, which means that it must contain two independent channels. The digitized signal must be placed in a separate array for each input. The STM32 peripherals are able to do this without the participation of the processor thanks to the presence of DMA.

The STM32 ADC has many modes of operation. For example, there are “regular” and “injected” channels. The point is that here you can form groups of channels, when the logical automaton will one by one make transformations in several channels and save the result. The difference between "regular" and "injected" channels is that in the first case the results are stored in one register, and in the second case - in separate registers for each channel. The second option is more convenient, but it has a limited number of channels, there are only 4. But here you need only 2, so this limitation is insignificant and you can use “injected” channels.

To configure a group of injected channels, there is a JSQR register, where the number of channels in the group (in this case, two) is encoded with two bits, and 4 more groups of 5 bits each encode the numbers of the ADC inputs involved in the group.

Moreover, if only 2 channels are involved, then they must be specified not in the JSQ1 and JSQ2 groups, but in the JSQ3 and JSQ4 groups. As if channels 3 and 4 of groups are working. However, data must be taken from JDR1 and JDR2, not JDR3 and JDR4. Likewise, the JOFR1 and JOFR2 registers should be used to offset the ADC code, not JOFR3 and JOFR4.

The conversion must be started strictly with the sampling frequency, for which the TIM2 timer and its TRGO event are used. The TRGO event is configured to fire on the Update event. In order for conversions to be carried out cyclically, SCAN mode is enabled. The sampling rate is 96 kHz, i.e. the period is 10.4 µs. During this time, the ADC must have time to make 2 conversions. The time taken by the ADC consists of two stages: the signal sampling and the conversion process. The conversion process takes 12.5 clock cycles. In this case, instead of the maximum frequency APB2 / 2 = 12 MHz, APB2 / 4 = 6 MHz was chosen, since there is enough time: the conversion will take a little more than 2 µs. The rest of the time can be spent sampling.

The sampling time value is quite critical, it imposes restrictions on the permissible output impedance of the signal source for the ADC. And, unexpectedly, the value of the admissible capacitance at the ADC input. In this case, it turned out to be possible to use the sampling duration of 13.5 clock cycles. As a result, the complete conversion will take 13.5 + 12.5 = 26 clock cycles.

The start interval must be at least one ADC cycle longer than the sequence length, so here the interval will be (26 + 26 + 1) * 1/6 [MHz] = 8.83 µs. This fits into a 10.4 µs sampling period, and without much ADC downtime.

At the end of the ADC conversion, it is possible to start a DMA cycle, which will transfer data from registers to memory. There is only one DMA channel connected to the ADC, so data can be taken from only one register (several “regular” channels can be involved) and added to one array. If several channels work, then the data in the array will be mixed, which is inconvenient for processing.

To take data from two registers of "injected" channels and store them in different arrays, two DMA channels are required. But they cannot be triggered by the ADC conversion end event, which is logical - it is impossible to understand in which channel the conversion ended. But since the conversion is started by a timer, then DMA can also be started using the same timer. After all, the ADC is a logical automaton, the conversion time for it is known and constant. It remains to assign two DMA channels and determine the moments in time for them when the ADC data will definitely be ready.

It is not known exactly how the conversion in the "injected" group of ADC channels takes place. There is no time diagram of the ADC operation in the documentation. Therefore, it is better to see in practice how the group is converted, at the same time to check whether the sequence fits into the sampling period. Observing the signal of the end of conversion of the ADC, it is impossible to determine which channel it belongs to. But everything can be done with a DAC. In a short cycle, you need to polling the timer and, at the time of starting the conversion of the group to the DAC, output the oscilloscope synchronization pulse. Then, in a loop, output the ADC output code to the DAC. Having fed a sinusoidal signal to the ADC input and changing its level, by changing the DAC output signal, you can clearly determine for which channel you see the data. Below are the moments of updating the ADC output register for the first and second channels.

ОСЦИЛЛОСКОП
САГА
СДЕЛАНО В СССР
0
1us
JDR1 update

Unfortunately, the speed of the DAC is too slow to observe such fast processes. The signal is somewhat delayed, but everything is understandable. The sampling period is 10.4 μs, which is just over 10 cells on the screen. It can be seen that the first channel finishes the transformation a little earlier than the middle of the screen, and the second one - a little earlier than the end of the screen. Those. the DMA channel for the first channel can be triggered in the middle of the sampling period, for the second - at the end.

DMA is used in cyclic mode, each channel works with its own array in memory. As a result, the data for its ADC input channel is updated in each of the arrays. It remains to take them from there on time for further processing. The first part of the data is processed by the half-buffer full flag, the second part is processed by the full buffer flag. The flags are polled in the main loop. The size of the buffer is chosen such that it takes just one sampling period of the "slow" domain to fill its half.

DC removal filter

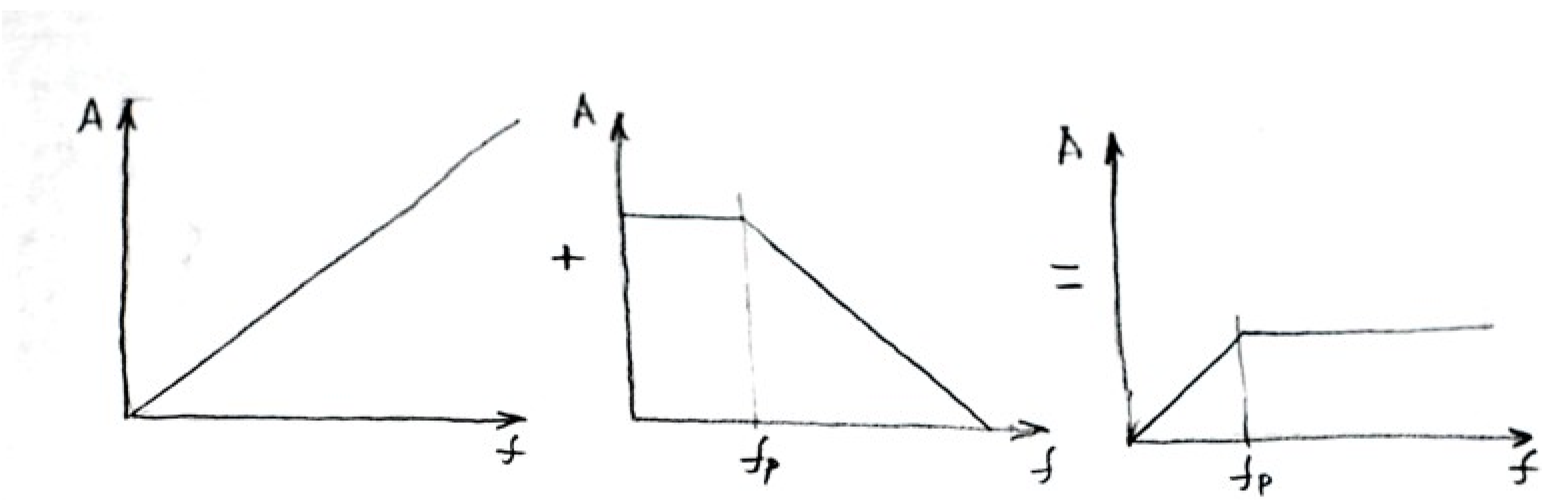
The first step is to remove the DC component from the signal. This must be done before sending a signal to the detector. It does not matter for the detector whether the input signal is constant or variable. Everything will go to the exit. With the dynamic range of the meter on the order of 60 dB, even such a small constant component as 0.1% of the ADC scale can be equal in magnitude to the useful signal and distort it. In practice, the constant component is much larger. The input level shifter only roughly shifts the signal by half the ADC scale. The circuit uses resistors with an accuracy of 1%, plus there are other errors.

Eliminating the DC component in the ADC dataset is a common task when digital signal processing is in progress. The simplest filter that suppresses the DC component is a differentiator. It has one zero at zero frequency. The DC gain is zero (which is required), and the gain increases with increasing frequency. For a discrete implementation of the differentiator, its difference equation has the following form:

y (n) = x (n) - x (n-1)

In fact, this is the simplest FIR filter. When implementing, take into account that the output value must have a capacity of at least 1 bit more than the input. Since the input signal, in theory, in one clock cycle can change over the entire scale.

A schematic of the frequency response of the differentiator is shown in the figure below (Fig. A). Such a course of the frequency response does not suit, since the transmission coefficient will change in the operating frequency range. We need to get a linear frequency response, and only below a certain cutoff frequency we need to have a falloff. Obviously, this requires adding a pole at the desired cutoff frequency. This can be done by connecting in series a differentiator and a first-order low-pass filter (its frequency response is shown in Fig. B). The combined frequency response will have the desired form (Fig. C).



This first-order low-pass filter is often referred to as a leaky integrator in these applications. Because in the analog version, such an integrator is implemented by adding a resistor in parallel with the capacitance, through which it will slowly discharge. For a discrete implementation, the difference equation of an ideal integrator is given below:

$$y(n) = y(n-1) + x(n)$$

To introduce a leak, the accumulated value must be multiplied by some factor less than one:

$$y(n) = A * y(n-1) + x(n), \quad 0 < A < 1$$

This is the simplest IIR low-pass filter, as it can also be called. The name "leaky integrator" is used to emphasize that this filter is applied in a somewhat unusual way - the operating frequency band lies in the stop band. Usually low-pass filter is used in a different way; bandwidth is used as the working one. But these are just the peculiarities of the terminology, which do not change the essence.

The cutoff frequency here should be chosen low compared to the sampling rate, which means that the factor A should be close to one. The cutoff frequency can be calculated using the formula:

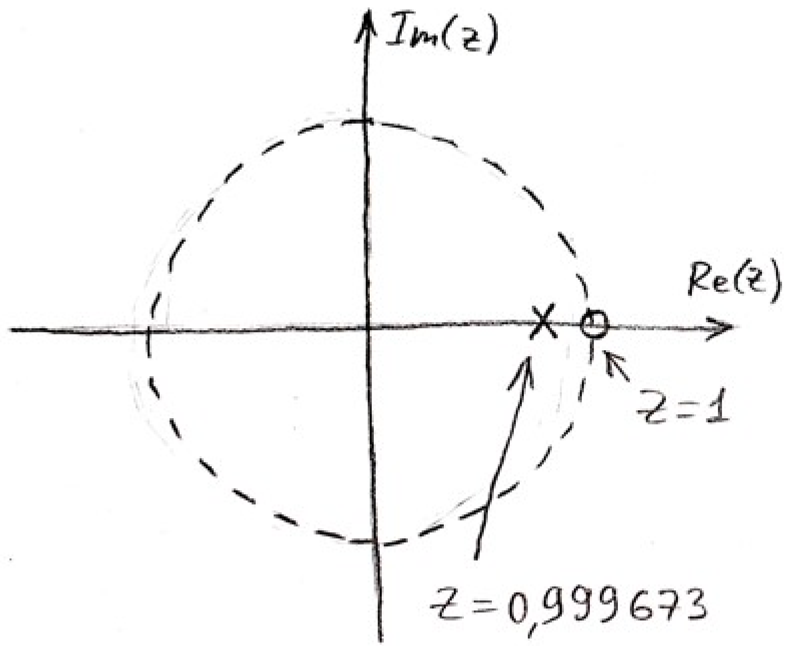
$f = (1 - A) * Fs / 2 * pi$, or $A = 1 - 2 * pi * f / Fs$, where Fs is the sampling rate.

The general difference equation for the filter will be as follows:

$$y(n) = x(n) - x(n-1) + A * y(n-1)$$

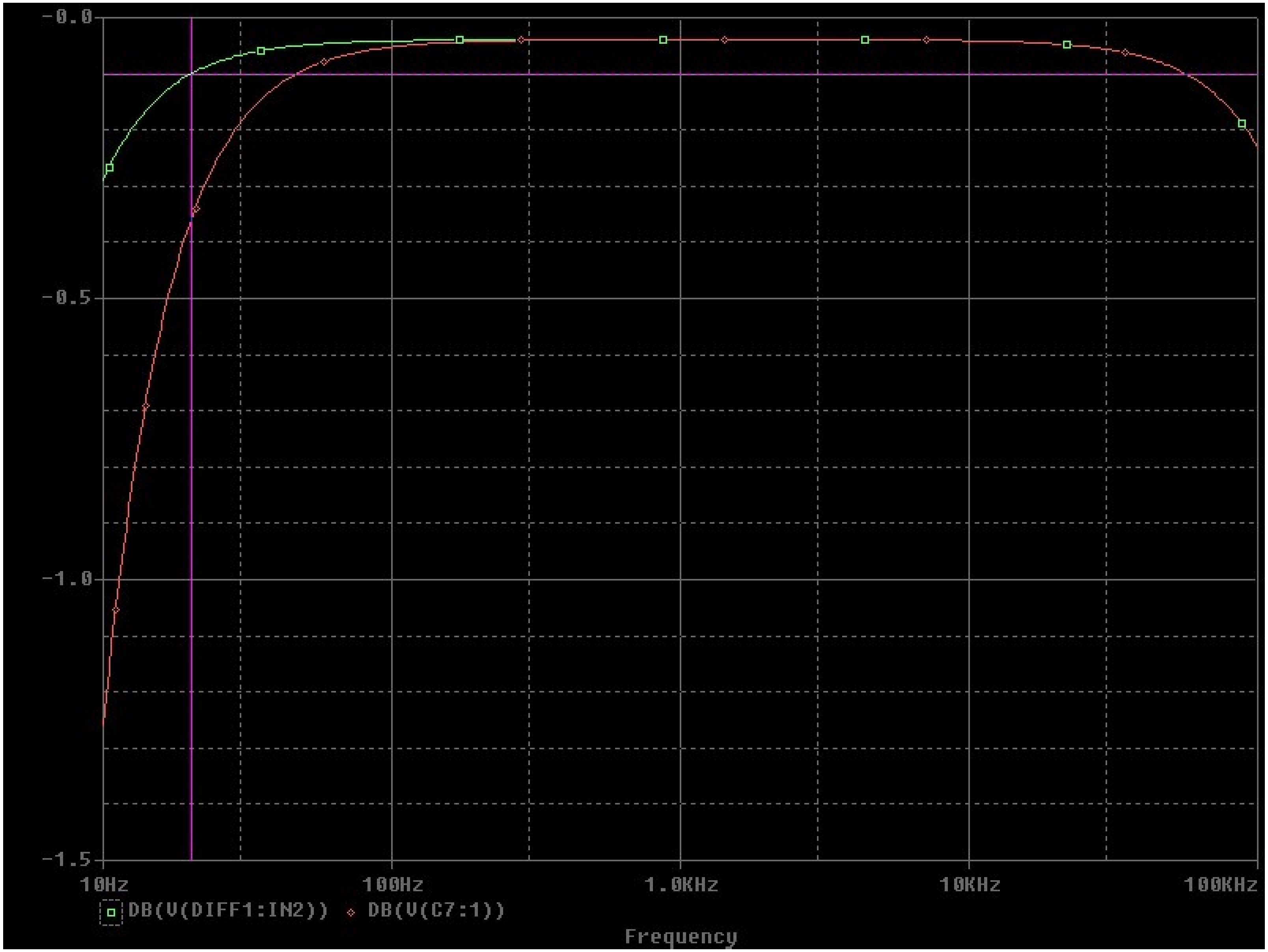
It is actually a first order IIR high-pass filter. It is equivalent to a differentiating RC chain.

Such a filter has one zero and one pole. If you depict them on the z-plane, then the zero will lie at the point $z = 1$, and a pole will lie slightly to the left of it on the axis.

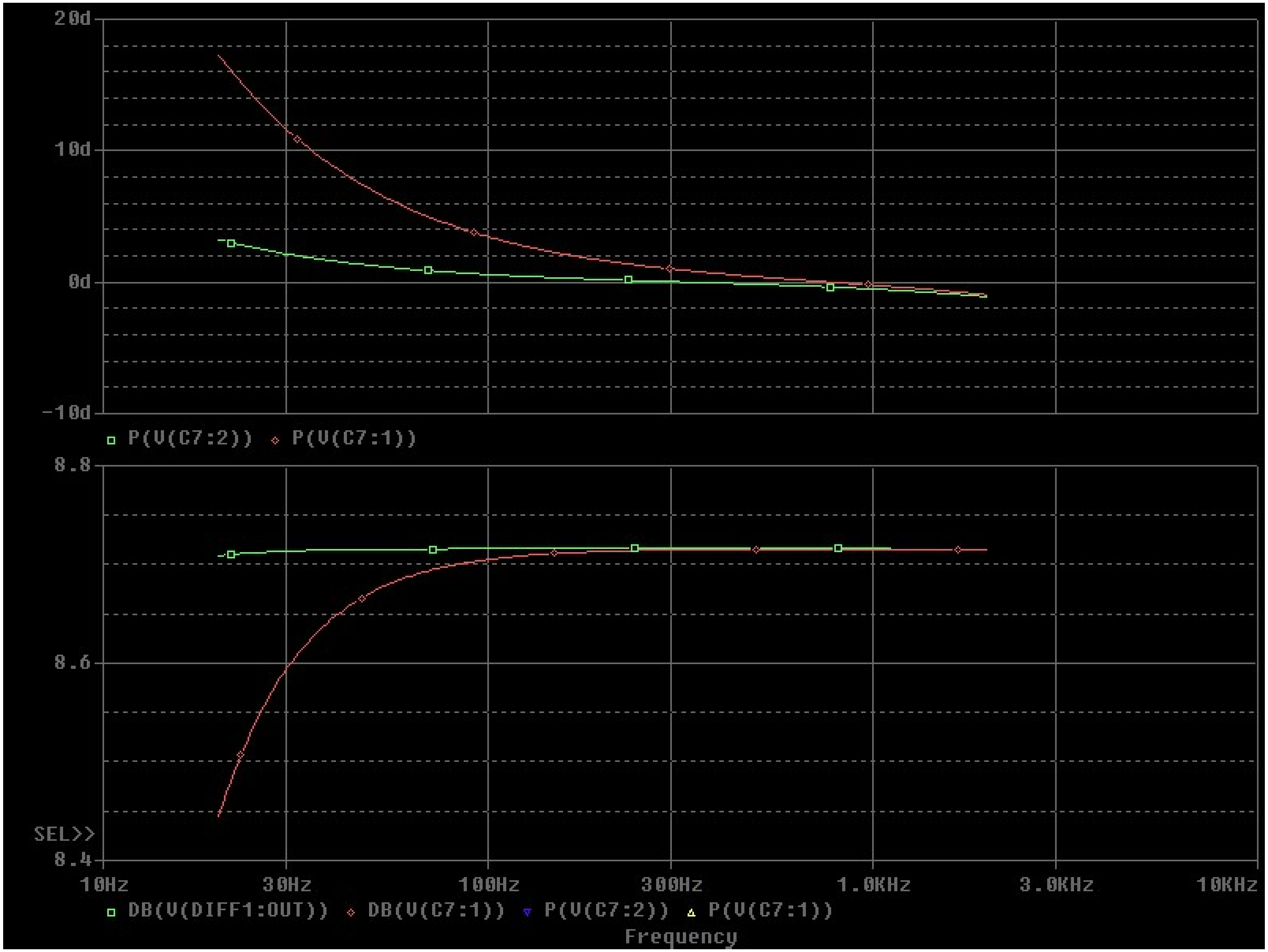


The audio frequency band starts at 20 Hz. But it is desirable to make the cutoff frequency lower so that a noticeable phase shift does not appear in the sound band. This filter is not phase-linear, so signal components of different frequencies will be delayed for different times. The result of their addition will give a different waveform, the peak level will be distorted. There are phase-linear FIR high-pass filters, but they require a lot of computational resources to obtain good linearity in the frequency response in the passband at a low cutoff frequency. There is a filter option for removing the constant component based on MAF (moving average filter). It doesn’t spend much computing power, but it requires a lot of memory to get a high sample rate to cutoff ratio.

In this case, the requirements for the filter are not very strict. It is necessary to remove the DC offset voltage, which can only change very slowly (for example, due to the temperature drift of the op-amp). A significant part of the offset to be compensated is generally constant, since it is associated with the divider error. Therefore, you can simply lower the cutoff frequency while minimizing the phase shift. A cutoff frequency of about 5 Hz is a good choice. The combined frequency response of the input buffer and buffer plus a high-pass filter with a cutoff frequency of 5 Hz is shown in the graph below.



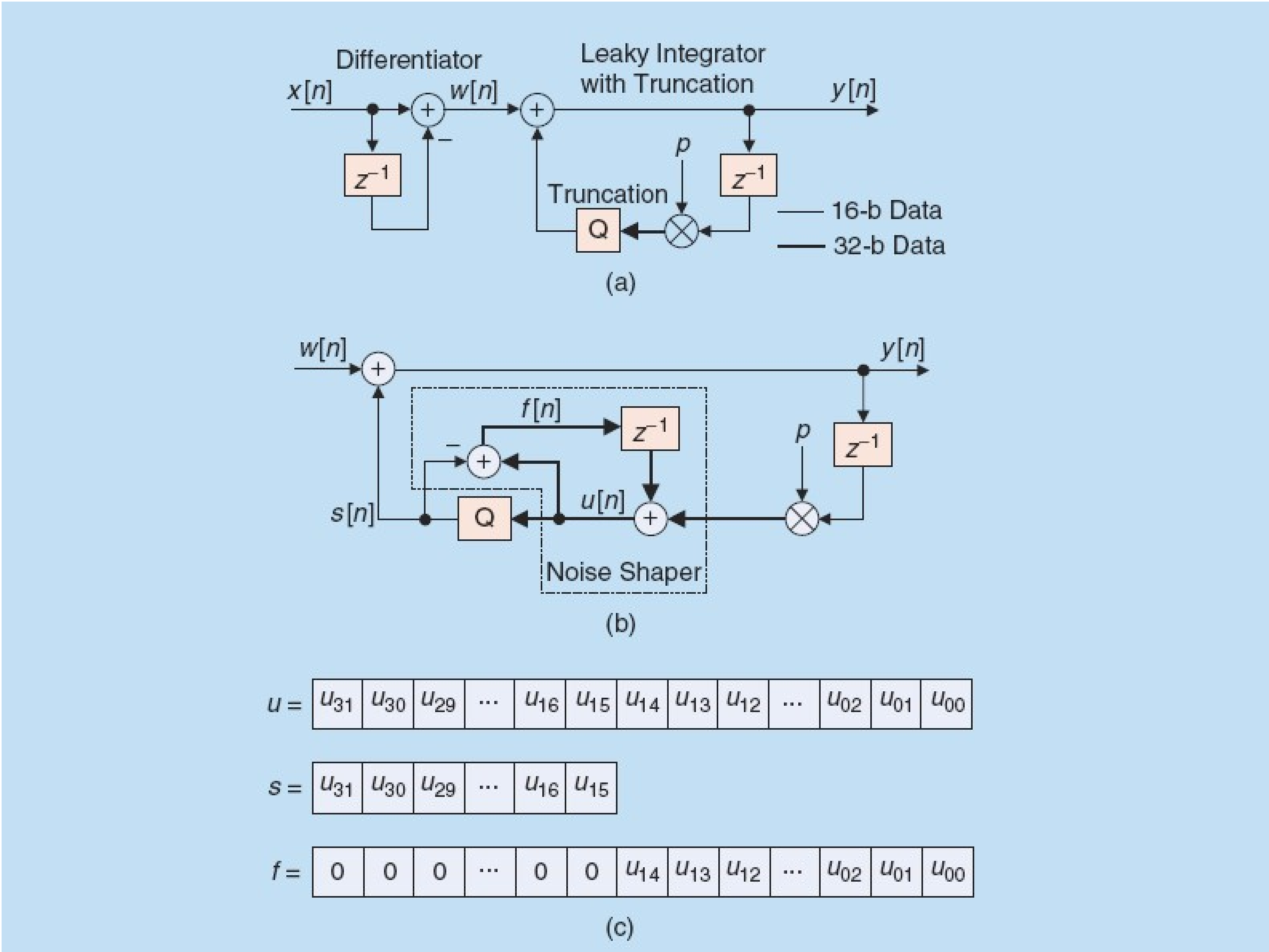
The phase shift at 20 Hz is about 17 degrees, which can be considered quite acceptable. The more detailed graph below shows the phase response (top) and frequency response (bottom) in the frequency range 20 Hz - 2 kHz.



The result is quite satisfying, you can get by with such a simple filter. It remains to implement it. If you use floating arithmetic, then there are no problems at all. But when implementing a filter in integer arithmetic, there are a number of problems. It is necessary to monitor the ranges of numbers at all stages of the calculation so that overflow does not occur. A differentiator and a leaky integrator can be connected in any order. But to avoid overflow, the differentiator should be turned on first.

Another problem, more insidious, is related to rounding errors. For a filter cutoff frequency of 5 Hz at a sampling rate of 96 kHz, the value of the coefficient A = 0.999673. To carry out calculations with such a coefficient with good accuracy, it is required to switch to increased bit depth. The reverse transition will represent quantization with a larger step, which leads to the appearance of an error, or quantization noise. As a result of the action of such an error in the filter, a parasitic constant component may appear at the output, which is even greater than the one that we suppress. This error can render the filter inoperative. As a practical test has shown, the filter does not actually work without correcting the requantization error.

In this case, the ADC is 12-bit, the input and output samples are conservatively placed in 16-bit signed integers. The stock is needed here for overflow protection. The multiplication will be done on a 32-bit grid. Then, when going from a 32-bit intermediate result to a 16-bit one, a quantization error will occur. In one of the publications, it is proposed to supplement the filter with a quantization noise spectrum shaper by conducting feedback by error.



In outwardly frightening form, the implementation of this method turns out to be very simple. I made my own implementation which uses the same error correction principle.

```
static const double POLE = 0.999673;
static const int32_t A = (int32_t) (UINT16_MAX * POLE);
```

```
static int32_t acc = 0;
static int16_t xx = 0;
static int16_t yy = 0;

x = Input;
acc = LO_W (acc) + A * yy;
yy = x - xx + HI_W (acc);
xx = x;
Output = yy;
```

Quantization occurs by discarding the lower half of a 32-bit number. The dropped value is a quantization error. The original number is signed, but the error is always positive. For example, if the lower digits of a positive number are replaced with zeros, the number will decrease; to correct the error, you will need to add some positive number. If a negative number is replaced with zeros, the number becomes modulo large, remaining negative. This means that to correct the error, you will again need to add some positive number. Therefore, the error can be extracted from the original 32-bit number, simply by zeroing the most significant half of it together with the sign bit.

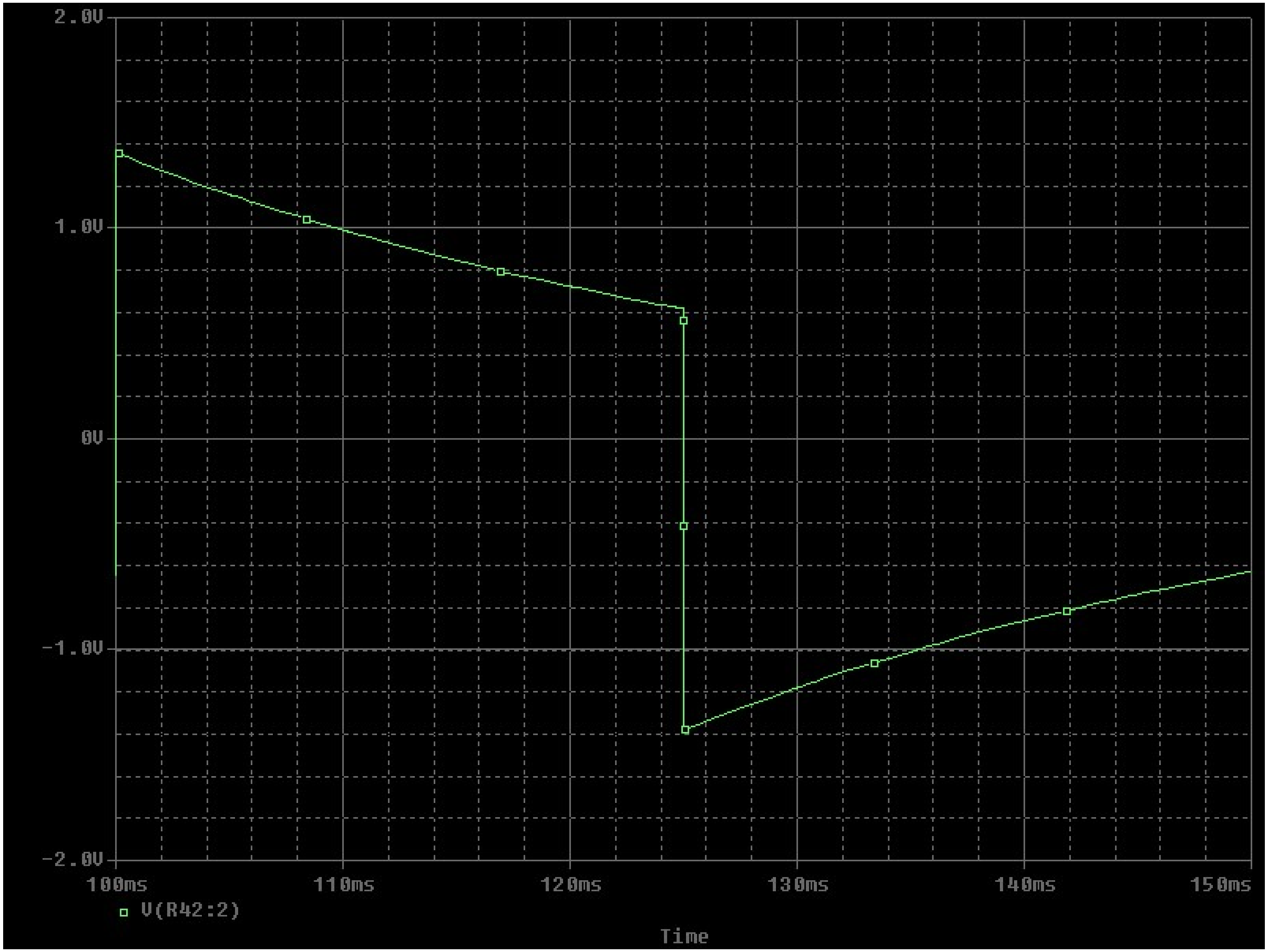
To check digital signal processing, it is convenient to use a DAC, then with an oscilloscope you can see what is actually happening with the signal. It is necessary to output readings to the DAC at the same rate as the ADC. But this is difficult to do when the buffer is filled using DMA. For debugging, I had to temporarily organize an interrupt from the timer, which used to start DMA. Instead of starting the DMA, an interrupt handler is executed in which the ADC is read, processed, and outputted to the DAC.

The first check is just a bunch of ADC-DAC with direct output to the DAC of the ADC codes. The ADC input receives a half-scale offset and a signal from the generator output. This is how the 5 kHz signal at the DAC output looks like:

To check the DC filter, a lower-frequency signal must be applied to the input, since a high-pass filter with a cutoff frequency of 5 Hz will practically have no effect on a 5 kHz signal. For clarity, I applied a meander with a frequency of 20 Hz to the input. This is how it looks at the DAC output without a filter:

If now included in the filter, the picture looks like this:

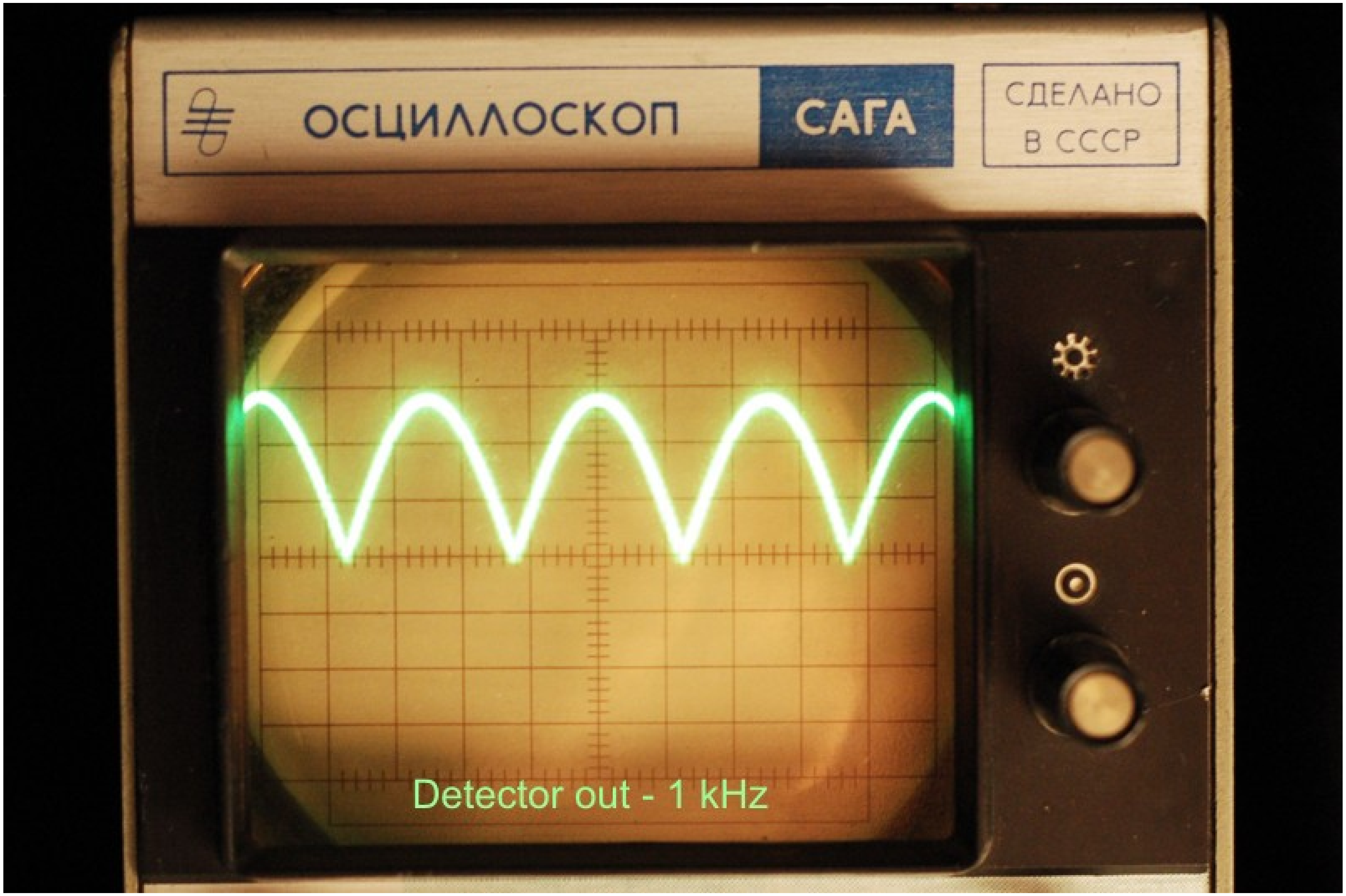
This results in a typical signal waveform for an RC differentiating circuit. To make sure that the filter works correctly, you need to compare the shape with that obtained on the model. This filter has an analog prototype, its model in PSpice gives an identical result:



If you change the constant offset at the ADC input, then the zero line at the DAC output momentarily goes away, then returns to its place again. The filter works as expected. The input data from the ADC is 12 bits wide. After the filter, the data has the same bit depth. To calculate this filter, the processor spends about 1.4 μs.

Detector

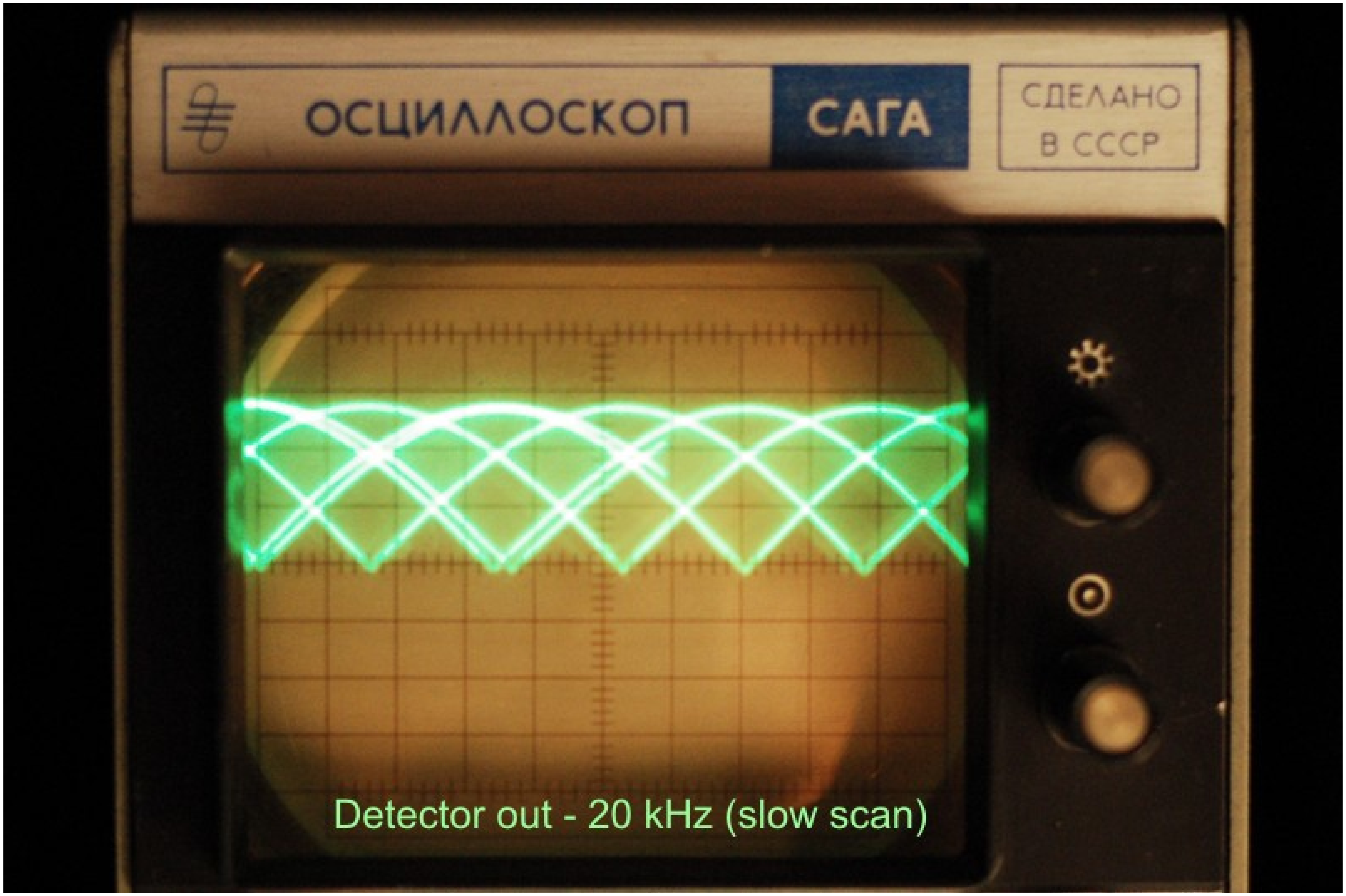
The detector is the simplest part of the processing. When the DC component is excluded from the signal, the task is reduced to calculating the absolute value. I also checked the detector's operation using a DAC.


ОСЦИЛЛОСКОП
САГА
СДЕЛАНО
В СССР
Detector out - 1 kHz


ОСЦИЛЛОСКОП
САГА
СДЕЛАНО
В СССР
Detector out - 5 kHz


ОСЦИЛЛОСКОП
САГА
СДЕЛАНО
В СССР
Detector out - 10 kHz

Even at a sampling rate of 96 kHz, the 20 kHz signal looks so-so. But if you turn on a slower sweep on the oscilloscope, the sinusoidal envelope is clearly visible.



At the output of the detector, the signal takes only positive values, so the sign bit is always zero. The data at the output of the detector becomes 11-bit.

Integration time

To obtain the specified integration time, the detector output must be smoothed with a filter. The time constant can be set arbitrarily, for example, you can take the value for a standard quasi-peak meter in accordance with GOST 21185-75 or IEC 60268-10.

An anti-aliasing filter is not a simple low-pass filter, but a more complex filter with different charge and discharge time constants. Charging must occur quickly to ensure the specified integration time of the meter.

According to GOST 21185-75, the readings should reach a level of -2 dB within 5 ms after the start of the 5 kHz test signal burst. This does not mean that the RC time constant should be 5ms. It actually takes a time constant of about 1.25ms. The difference equation of the filter is as follows:

y (n) = y (n-1) + A * (x (n) - y (n-1)), where A = 0.0083 for Fs = 96 kHz

This filter turns on only when the signal rises, the signal is held at the same level during the fall. A separate filter will be responsible for the signal decay. In fact, two IIR filters work here, which form the integration time to indicate the average (or quasi-peak) value, which is displayed by bars, and the peak value, which is displayed by a single burning segment - a dot. To obtain the required accuracy of calculations, the accumulation is carried out in 32-bit accumulators. The filter coefficients are 16-bit unsigned. They are defined as A = 65536/96000 / tau.

First, you can test this filter with the same charge and discharge times. To obtain the test signal, a field switch is used that modulates the 5 kHz signal at 20 Hz. The pause key does not completely suppress the signal, but this does not really matter.

At the output of the filter, a typical low-pass filter is observed. In 5 ms, the signal just reaches the required level, the time constant coincided with the calculated one.

Slow domain processing

The return time of the meter in the standard is defined as 1.7 seconds before the level drops by 20 dB. This requires a filter time constant of approximately 740 ms. This is too long to process in the 96 kHz domain. Therefore, further processing is carried out in the "slow" domain. The reduced sampling rate is formed as the frequency of filling 256-point arrays with ADC data, ie 96000/256 = 375 Hz. This is a quite suitable sampling rate for this filter, for 740 ms the filter factor is 0.0036, which is a perfectly acceptable value.

Processing in the "slow" domain begins with scaling the signal, which is necessary to align the readings of the bars and points on the stationary signal at different integration and retrace time constants. The 11-bit output of the previous filters is multiplied by the scaling factors. The output is a maximum code of 10000, this is the range that is used in the lookup table. After multiplying by the coefficients, the code is limited to about 10% above the selected maximum code. At maximum values, the gain is 32 (approximately +30 dB). You can use this gain to shift the scale when you need a more accurate measurement of small signals.

Response time

Level meters have as many as 4 dynamic characteristics:

Различают следующие четыре динамические характеристики ИУ:

Время интеграции $t_и$ — длительность одиночного радиоимпульса, при которой указатель показывающего прибора доходит до отметки, лежащей на 2 дБ ниже показания на непрерывном гармоническом сигнале, имеющем частоту и амплитуду сигнала заполнения радиоимпульса.

Время срабатывания указателя $t_{ср}$ — интервал времени между моментом подачи непрерывного гармонического сигнала частоты 1000 Гц номинального уровня на вход ИУ и тем моментом, когда указатель показывающего прибора доходит до отметки —1 дБ.

Время возврата указателя $t_в$ — интервал времени между моментом выключения непрерывного гармонического сигнала частоты 1000 Гц номинального уровня на входе ИУ и тем моментом, когда указатель показывающего прибора доходит до отметки —20 дБ (10%).

Переброс указателя δ — разность между максимальным показанием при скачкообразной подаче непрерывного гармонического сигнала на вход ИУ и его показанием в стационарном режиме, то есть после окончания процесса успокоения подвижной системы. Переброс выражается в децибелах или процентах относительно показания в стационарном режиме.

The integration time and the return (return) time are clear. The pointer flip is a characteristic of the pointer devices, it is not relevant for the LED ruler. But the response time is quite relevant.

Показывающие приборы неэлектромеханической системы могут иметь гораздо меньшие значения времени срабатывания (менее 1 мс). В связи с этим такие приборы называют *безынерционными*. При этом точный отсчет уровня затрудняется. Поэтому для них рекомендуется как бы имитировать время срабатывания, осуществляя задержку сигналов на 100—200 мс.

For inertialess devices, ballistics are artificially formed, similar to switch devices. An increase in the forward travel time should not increase the integration time of the meter. Short signal pulses should still display correctly. To do this, the detector must "wait" for the reading to reach the measured value, and only then start the reverse run. In industrial analog meters (for example, IU-12), rather complex circuits were used for this.

If we compare the work of different LED level meters, it turns out that limiting the rate of growth of the length of the ruler has a very positive effect on perception. In amateur designs of quasi-peak indicators, when the signal level rises sharply, the bar immediately jumps from one length to another. There is no such thing in professional quasi-peak meters. There is no such thing in VU meters, there, due to the long integration time, smoothness is obtained by itself.

> Индикатор на светодиодах практически безынерционный. Поэтому на выходе квазипикового детектора необходимо предусмотреть интегрирующую цепочку, обеспечивающую требуемое время срабатывания. В соответствии с определением $t_{\text{ср}}$ постоянная времени этой цепи равна $\tau_{\text{зд}} = 0{,}434\, t_{\text{ср}} = 0{,}434 \cdot 0{,}2 = 0{,}087$ с.

To form the response time, two IIR filters are used, which work in the "slow" domain. The output code of these filters is directly related to the meter readings. To obtain the required accuracy of calculations, the accumulation is carried out in 32-bit accumulators. Like integration time filters, these filters only work on rising signals. The filter coefficients are 16-bit unsigned. They are defined as A = 65536/375 / tau.

Trigger time is defined as the interval from the moment the test tone bursts to the moment when the reading reaches -1 dB. Therefore, when the output code of the trigger filter reaches the input code with an accuracy of 1 dB, the trigger interval ends.

Hold and retrace time

It was said above that the integration filters only work when the signal rises. They can only increase the output code. Special filters are responsible for the signal decay, which act on the accumulators of the integration filters. They turn on after the end of the response interval. Since the signal roll-off is slow, these filters operate in a "slow" domain. The filter coefficients are 16-bit unsigned. They are defined as A = 65536/375 / tau.

A drop in the integration filters does not mean a drop in the meter reading. The readings are associated with the pickup filters and may have some hold time when the readings remain unchanged after the peak of the signal. This time is set using special coefficients, which are calculated by the formula A = 375 * t_hold. If you set a zero value, the readings will not be held, immediately after the end of the response interval, the reverse will begin.

When the hold time ends, the meter begins to reverse. For this, return filters are used. These filters act on the trip filter batteries. Filters work on a "slow" domain. Reverse filter coefficients are 16-bit unsigned. They are defined as A = 65536/375 / tau. This time constant determines the return time of the meter.

Separate flybacks for the trigger filters allow for reduced flyback times for the integration filters, which reduces the effect of the time between signal peaks on the meter reading. And there is such an effect - if the signal peaks at the meter's input are repeated more often, then the readings increase (despite the fact that the magnitude of the peaks themselves remains unchanged). This is due to the fact that by the time the next peak begins, the integration filter did not have time to completely "discharge", a new increase in its output signal begins with a higher initial value. As a result, during the duration of the peak, it will be able to reach a greater value than in the case of rarely repeated peaks.

When the filter has different integration and flyback time constants, the magnitude of the output signal will depend on the ratio of these times. If these times were the same, it would be a simple averaging filter that would output the average of the signal (for sine, this is 0.637 of the peak value). If you increase the retrace time with respect to the integration time, the output will increase and at the limit will be equal to the peak value of the input signal when the ratio of the retrace time to the integration time becomes infinite. The situation is similar with a conventional analog detector with a smoothing filter. Below is a graph of the dependence of the output voltage of such a detector on the ratio of the charging and discharging resistors of the filter capacitor.Only here it is necessary to take into accountthat both resistors always work and form a divider, therefore, with equal ratings, the output signal is halved.

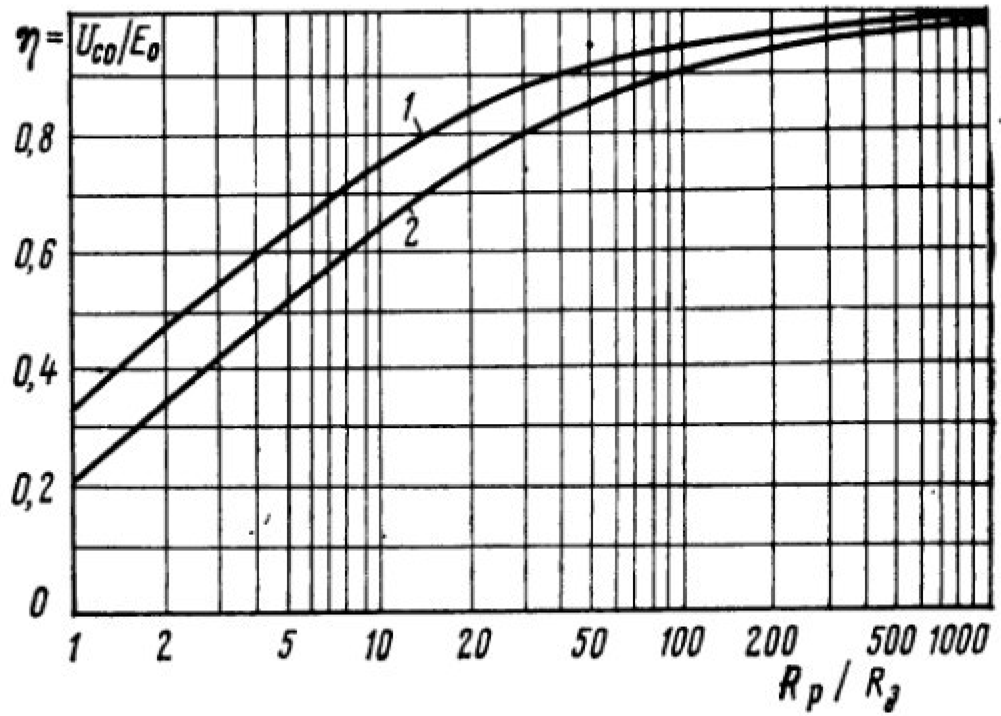


Рис. 5.5 Зависимость напряжения на конденсаторе фильтра от соотношения сопротивлений цепей заряда и разряда:

1 — для двухполупериодного детектора; 2 — для однополупериодного детектора

Therefore, changing the ratio of the time constants will violate the calibration of the meter, which requires correction using the scaling factors.

The maximum value of the measured signal can be memorized and held for some time. The accumulators of the maximum value are used for storing. The holding time is set using coefficients that are calculated by the formula A = 375 * t_hold. If you set the value to zero, there will be no hold of the reading, immediately after the peak of the signal, the reverse will begin.

When the holding time ends, the reversal of the maximum values begins. The return filter acts on the accumulators of the maximum values. It runs on a "slow" domain. The retraction time constants are determined by coefficients that are calculated by the formula A = 65536/375 / tau.

In parallel with the accumulators of the maximum values, one more accumulator operates. They are used to collect statistics.

When the meter is in operation, the values in these batteries can only increase. The reverse is not provided here. During operation, the maximum measured value is accumulated. These batteries are manually reset using a special command.

Signal processing is performed in the same way for the average (or quasi-peak) value, which is displayed in bars, and for the peak value, which is displayed as a single burning segment - a dot. The difference is determined only by the values of the coefficients.

For a standard quasi-peak meter, the integration tau is set to 1.25 ms (5 ms rise to -2 dB), the retrace tau 739 ms (down to -20 dB in 1700 ms), the trigger tau 43 - 87 ms (100 - 200 ms rise to -1 dB), hold time - 0 ms.

For the peak meter, the integration tau is set to 0 ms, the actuation tau is 0 to 87 ms, the hold time is about 2000 ms, the retrace tau is from 0 ms (the point disappears immediately) to 739 ms (the point smoothly decreases in 1700 ms to the level of -20 dB). The level scaling factor can be set to zero to completely turn off the peak indication.

VU mode

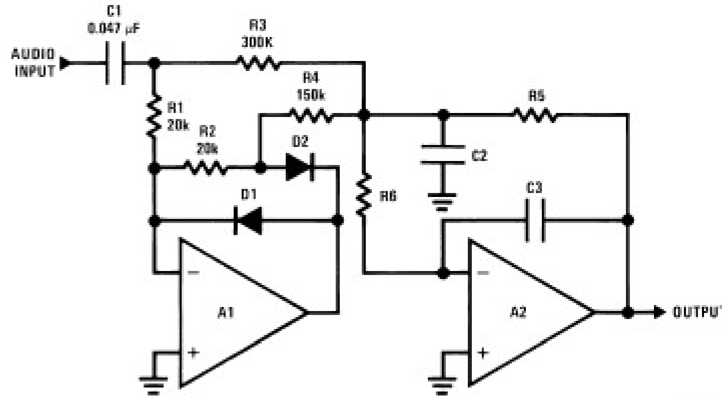
Previously, the most common type of meter was the VU. To be able to compare readings with old devices, it would be useful to have such a mode. For the VU meter, the standard specifies the same charge and discharge time constants of 300 ms. At first glance, it seems that this problem can be solved simply - it is enough to choose the required integration and return time constants. However, for VU meters, the standard defines a time of 300 ms as the time it takes for the reading to reach 99% of steady state, followed by an overshoot of 1 to 1.5%. Previously, this was provided by a mechanical moving system of the measuring instrument. To do this electrically, a 2nd order low-pass filter with a cutoff frequency of 2.1 Hz and a Q of 0.62 is required. This is something between the Bessel and Butterworth filters.It was possible to calculate the analog prototype using the FilterPro program, which allows you to set the quality factor of the filter links. But I could not find a suitable software for calculating such a digital filter.

If you look at the VU meter circuits where an analog detector is used, then a second-order low-pass filter with the specified characteristics is necessarily installed after it. For example, it can be found in the datasheet on LM3916. There, this filter is made based on the MFB structure:

VU Meter

The audio level meter most frequently encountered is the VU meter. Its characteristics are defined as the ANSI specification C165. The LM3916's outputs correspond to the meter indications specified with the omission of the −2 VU indication. The VU scale divisions differ slightly from a linear scale in order to obtain whole numbers in dB.

Some of the most important specifications for an AC meter are its dynamic characteristics. These define how the meter responds to transients and how fast the reading decays. The VU meter is a relatively slow full-wave averaging type, specified to reach 99% deflection in 300 ms and overshoot by 1 to 1.5%. In engineering terms this means a slightly underdamped second order response with a resonant frequency of 2.1 Hz and a Q of 0.62. *Figure 7* depicts a simple rectifier/filter circuit that meets these criteria.



Design Equations

$$\frac{1}{R5 \cdot R6 \cdot C2 \cdot C3} = \omega_0^{2} = 177\ \mathrm{sec}^{-2}$$

$$\frac{1}{C2}\left(\frac{1}{R3} + \frac{1}{R4} + \frac{1}{R5} + \frac{1}{R6}\right) = \frac{\omega_0}{Q} = 21.5\ \mathrm{sec}^{-1}$$

R3 = 2R4

R1 = R2 << R4

A1, A2: 1/2 LF353

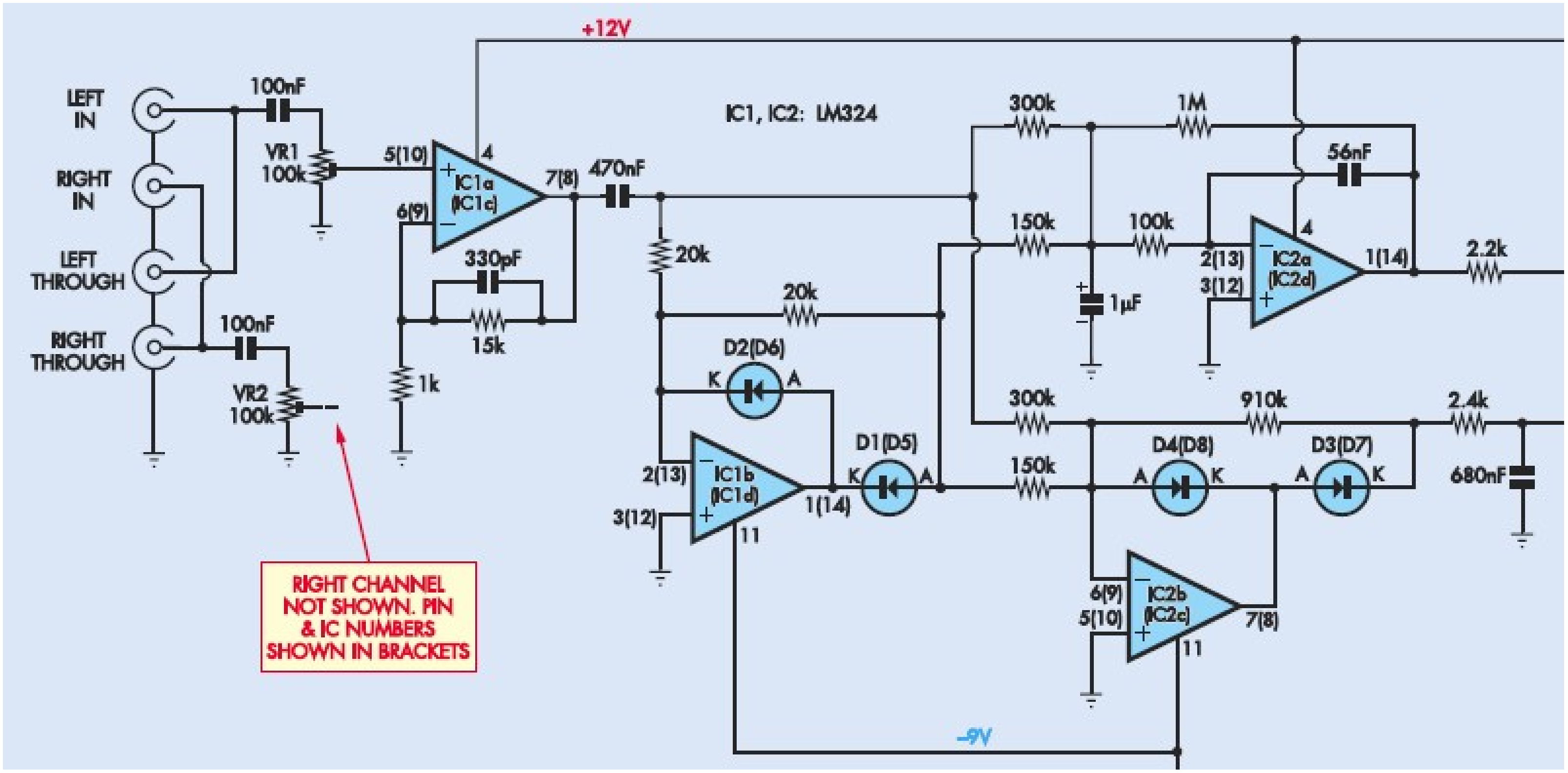
D1, D2: 1N914 OR 1N4148

*Reaches 99% level at 300 ms after applied tone burst and overshoots 1.2%.

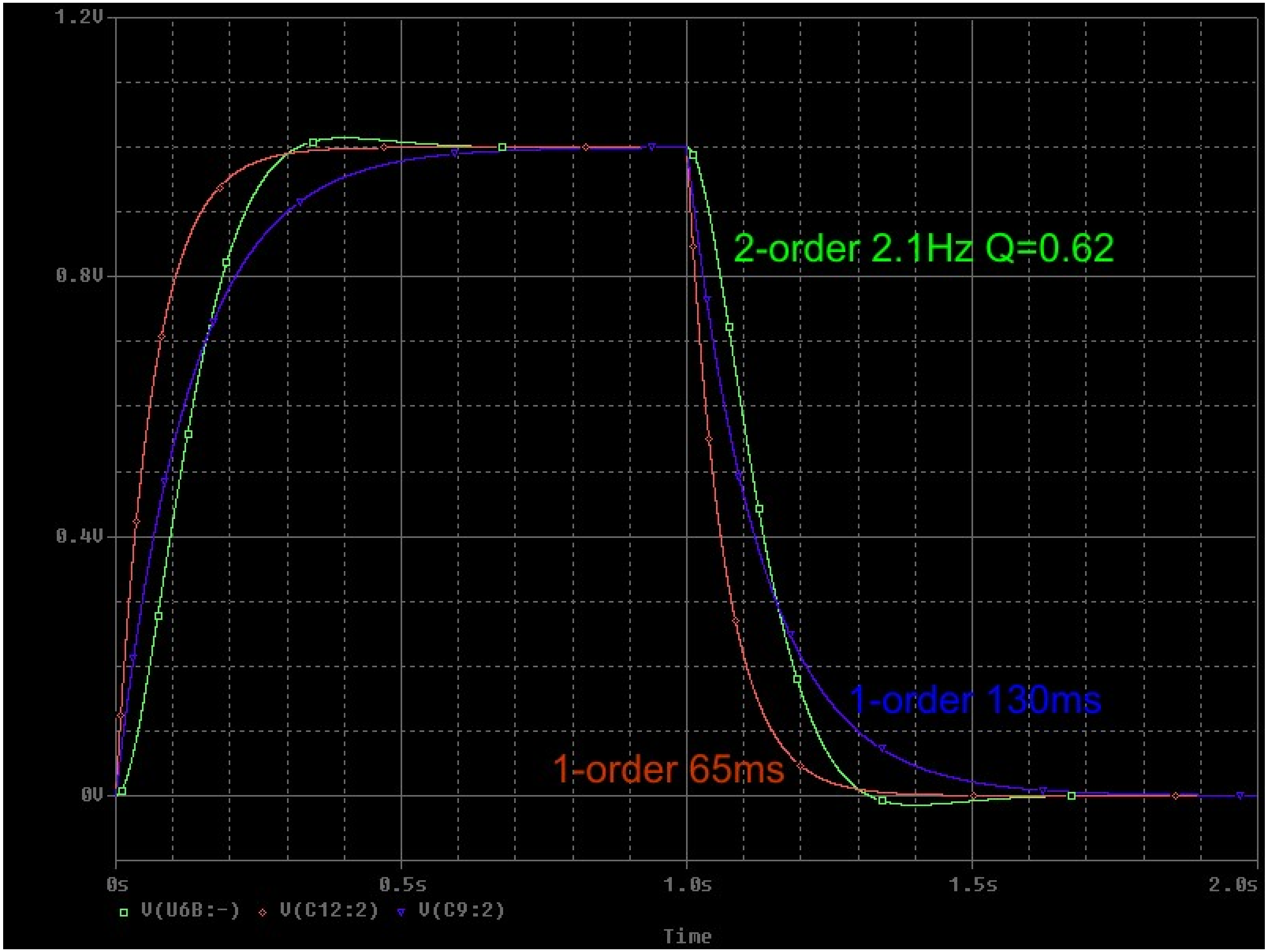
GAIN	R5	R6	C2	C3
1	100k	43k	2.0	0.56 µF
10	1M	100k	1.0	0.056 µF

FIGURE 7. Full-Wave Average Detector to VU Meter Specifications*

A similar circuit is used in the level meter from the magazine "Everyday Practical Electronics", March 2009:



On the other hand, I don't really want to introduce an extra filter, which is needed only in VU mode. Better to do without it. To get similar meter ballistics using a 1st order filter, you can try to find the time constant. If you want to set up to 99% in 300ms, as required by the standard, then 65ms tau will be needed (red graph).



It can be seen that the slew rate of the output signal of such a filter is greatly overestimated, the meter will greatly overestimate the readings for short signal flashes. You can select the transient response on average similar to the standard for VU. This will take a tau of approximately 130ms (blue graph). Then, in 300 ms, only 90% will be reached, but in general the meter's behavior will be very close to VU. Although in this case it is not entirely correct to claim that the meter supports VU mode, in practice the difference in readings will be insignificant.

Additional filters

Digital signal processing makes it possible to implement additional modes of the meter, in addition to measuring the peak and quasi-peak levels.

The RMS level can be displayed. But as mentioned above, this is important for assessing the loudness level, but for an analog tape recorder, this does not make much sense. Implementation of such a mode is possible, but it may require the use of a microcontroller with a higher speed.

More modern algorithms for measuring loudness level (in LUFS) use two 2nd order weighting filters, the coefficients of which are given in the ITU-R BS.1770-4 standard. Such filters can also be implemented if necessary.

Another mode with frequency weighting is the implementation of a correction similar to the frequency response of an ultrasonic tape recorder. Then you can see the real overload capacity, taking into account the spectral composition of the signal. The weighting filter can have a certain averaged characteristic of the recording channel.

Another additional function of the meter can be the extraction of the signal of the 3rd harmonic and measurement of its level. This will determine the maximum recording level for a particular tape. To implement this function, you need digital bandpass filters.

Tabular scale assignment

The result of digital processing of the input signal is two numbers (for each of the stereo channels): this is the current value of the signal for indication by a column and for indication by a dot. These numbers are updated at the sampling rate of the "slow" domain, i. E. 375 Hz. With this frequency, the output to the LEDs occurs. The nominal range of numbers is from 0 to 10000. They represent the magnitude of the signal on a linear scale. At the same time, the scale of the meter can be non-linear. It is often made logarithmic, or even more complex, for example, S-shaped with a stretched zone near 0 dB.

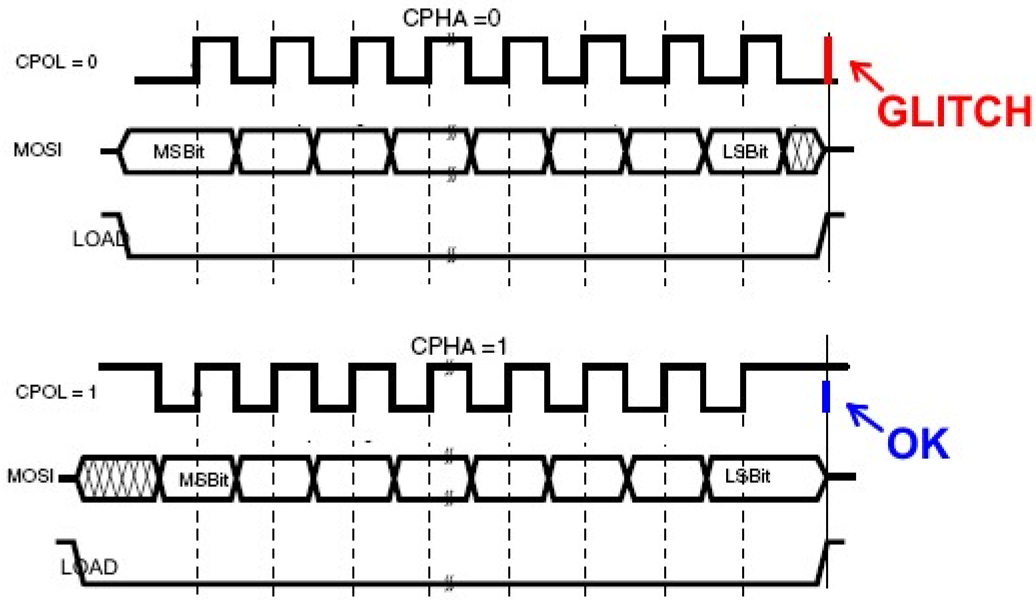
A table is used to set an arbitrary scale. It is common for two channels and contains the number of elements equal to the number of LEDs in the line. Each value in the table corresponds to the activation threshold of the corresponding segment of the ruler. The segment turns on if the signal code becomes greater than this value. For each segment, the actuation level is set in dB with a discreteness of 0.01 dB. Those. the level values in dB are entered in the table, multiplied by 100. The values in the table should be monotonic. The last item in the table is the maximum level. The difference between the maximum and minimum levels can be up to 80 dB.All levels below -80 dB relative to the maximum level are considered minus infinity. This level can be set for the first segment of the ruler so that it is constantly on.

Displaying values

The LED strips are connected to the outputs of the 74HC595 type shift registers, which are daisy-chained and loaded through the SPI port. The hardware SPI of the STM32 microcontroller is used. Since the exchange is unidirectional, the transmit-only mode is used. In this mode, there are features of checking the end of the transfer. Here is a quote from the documentation: “During discontinuous communications, there is a 2 APB clock period delay between the write operation to SPI_DR and the BSY bit setting. As a consequence, in transmit-only mode, it is mandatory to wait first until TXE is set and then until BSY is cleared after writing the last data. "Thus, analyzing only the BSY flag, you can erroneously generate a LOAD signal at the beginning of the transmission of the last byte, and not after it ends. Therefore,it is imperative to check the TXE flag.

```
while (! (SPI1-> SR & SPI_SR_TXE));
while (SPI1-> SR & SPI_SR_BSY);
```

A very long chain of shift registers is used here, each of which is loaded onto LEDs. Only in the lines there are 100 LEDs, not counting the backlight and additional stencils. When loading new data into registers on the LOAD signal, a large number of register outputs can be switched simultaneously, which can lead to noise on the register control lines. In this case, this interference has the form of a short positive pulse at the moment when the edge of the LOAD signal appears. The SCLK clock signal line is the most sensitive to this; interference can cause unnecessary shift and distortion of information. To avoid the influence of noise on this line, the SPI operating mode CPHA = 1 and CPOL = 1 is used, when at the end of the data transfer the SPI line is set to HIGH.



For the convenience of wiring the printed circuit board, the segments of the indicators are chaotically connected to the registers. This required additional software effort. I had to create a large array of constants and fill it with structures in the form of pairs of numbers "Register number - output number". If it is necessary to light the segment with the number N, a pair of numbers is taken from this array at the index N - the register number and the number of the output to which this segment is connected. First, a copy of the contents of all registers in memory is formed, then it is loaded into the registers.

```
const Seg_t SegsL [SEGS] [CHANS] =
{
  {{U1, Q2}, / * segment L01 * / {U1, Q7}, / * segment R01 * /},
  {{U1, Q1}, / * segment L02 * / {U1, Q6}, / * segment R02 * /},
  {{U2, Q0}, / * segment L03 * / {U2, Q4}, / * segment R03 * /},
```

```
{{U2, Q3}, / * segment L04 * / {U2, Q5}, / * segment R04 * /},
{{U2, Q2}, / * segment L05 * / {U2, Q7}, / * segment R05 * /},
{{U2, Q1}, / * segment L06 * / {U2, Q6}, / * segment R06 * /},
{{U3, Q0}, / * segment L07 * / {U3, Q4}, / * segment R07 * /},
{{U3, Q3}, / * segment L08 * / {U3, Q5}, / * segment R08 * /},

................................ etc.
```

To access the segments, several methods are implemented: you can light a column of a given length at once, or a separate segment. Stencil and scale LEDs are individually controlled.

After the table recoding, a positional code is formed, according to which a set of burning segments is formed. After this set is formed in memory, it is loaded into registers. This happens periodically, with a frequency of 375 Hz.

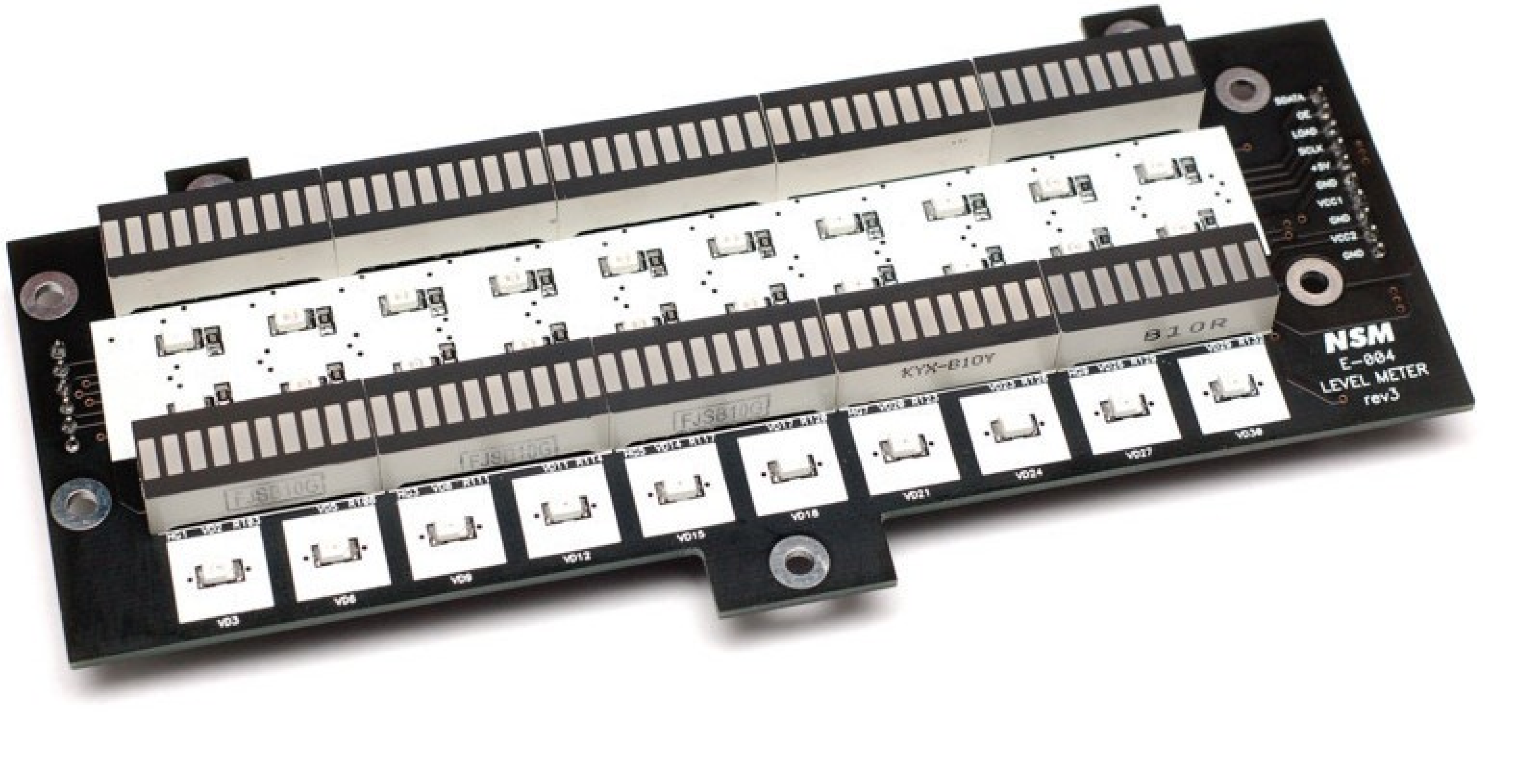
Presets

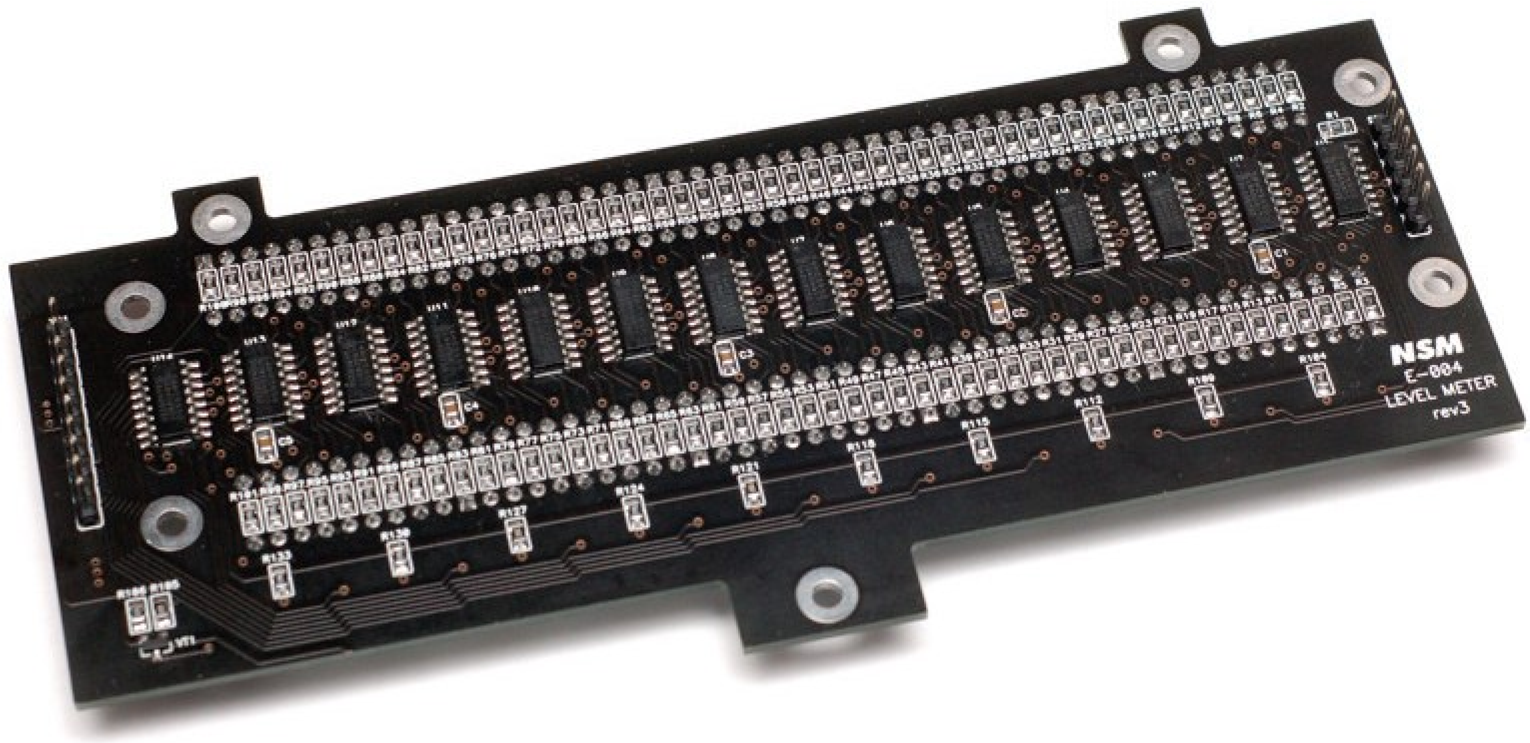
Filter coefficients, scaling factors, look-up tables, and other parameters are stored in non-volatile memory. In total, 4 sets of parameters are saved, which are presets of operating modes for the level meter. Presets can be switched using jumpers on the board. A variant of the firmware is possible, where the presets are switched "in a circle" using a button connected instead of one of the jumpers. Presets can also be switched using commands from the control processor via the RS-485 interface. In addition, this interface has the ability to change any of the parameters separately.

Meter design

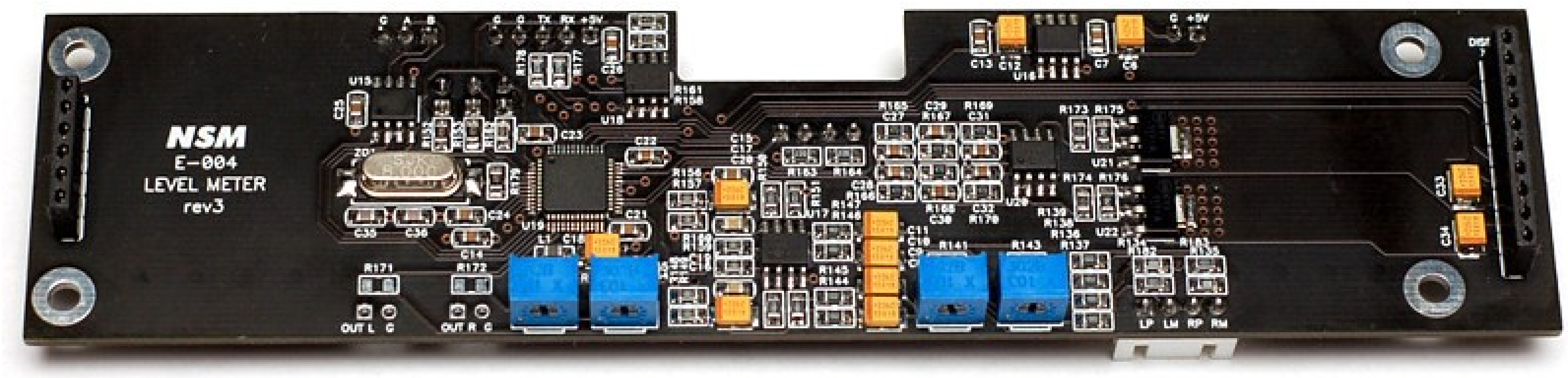
Structurally, the meter consists of two printed circuit boards - an indication board and a processor board.

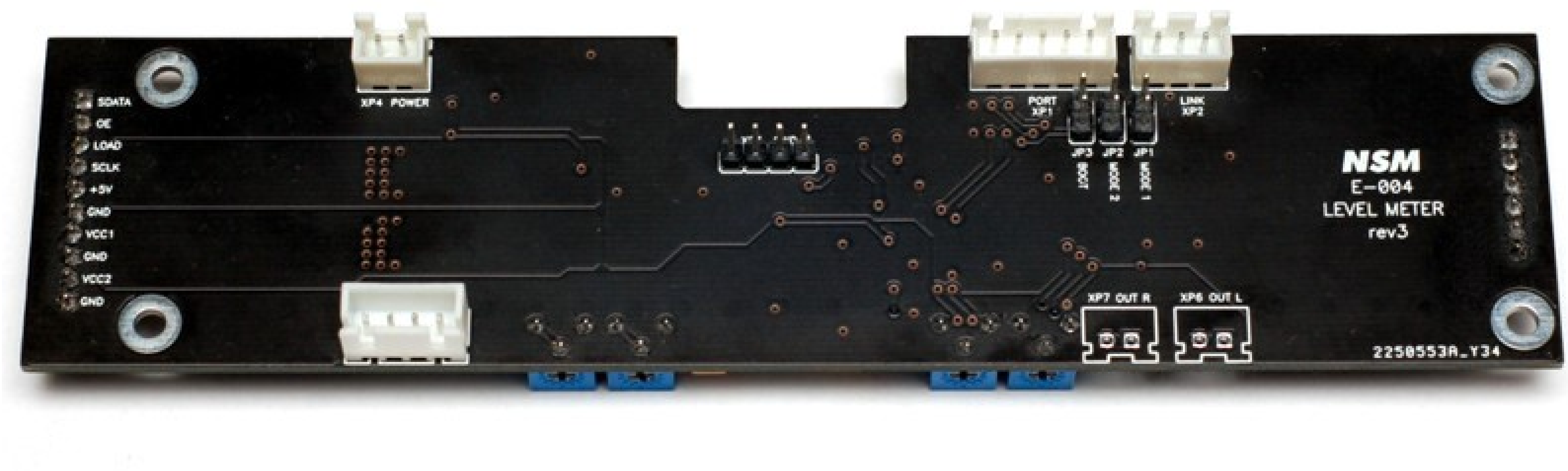
The display board contains LED bars, backlight LEDs, stencil LEDs, and shift registers. Each segment and LED has a limiting resistor. There is a transistor switch to turn the backlight on or off. To attenuate stray light reflections from the board, the board mask is made black. At the locations of the backlight LEDs, white rectangles are applied in the silk-screen layer, which serve as light reflectors.


NSM
E-004
LEVEL METER
rev3
KYX-B10Y
B10R



The processor board contains a microcontroller, an input amplifier, a +3.3 V voltage regulator, a nonvolatile memory chip, an RS-485 interface chip, as well as two adjustable stabilizers to adjust the brightness of the rulers and the scale backlight. Setting the brightness can be done both programmatically and using trimming resistors. In the case of software brightness control, it is possible to use the built-in DAC or PWM. When using PWM, it is smoothed using active low-pass filters, LEDs are always supplied with direct current. Two more trimming resistors are used to adjust the input signal level. On the other side of the board there is a power connector, an input connector, as well as connectors for a communication port with a computer and an RS-485 port. Additionally,analog output connectors can be installed to control dial indicators. The board also has pins for the SWD debug connector, as well as pins for setting the BOOT jumper and preset selection jumpers.





The boards are connected to each other using the XS1 connector (PLS / PBS), which has 10 pins. On the other side of the board, a similar 6-pin connector is installed, it only has the function of fastening, it does not make any electrical connections. The boards are fastened using four M3 threaded racks 10.5 - 12 mm high.

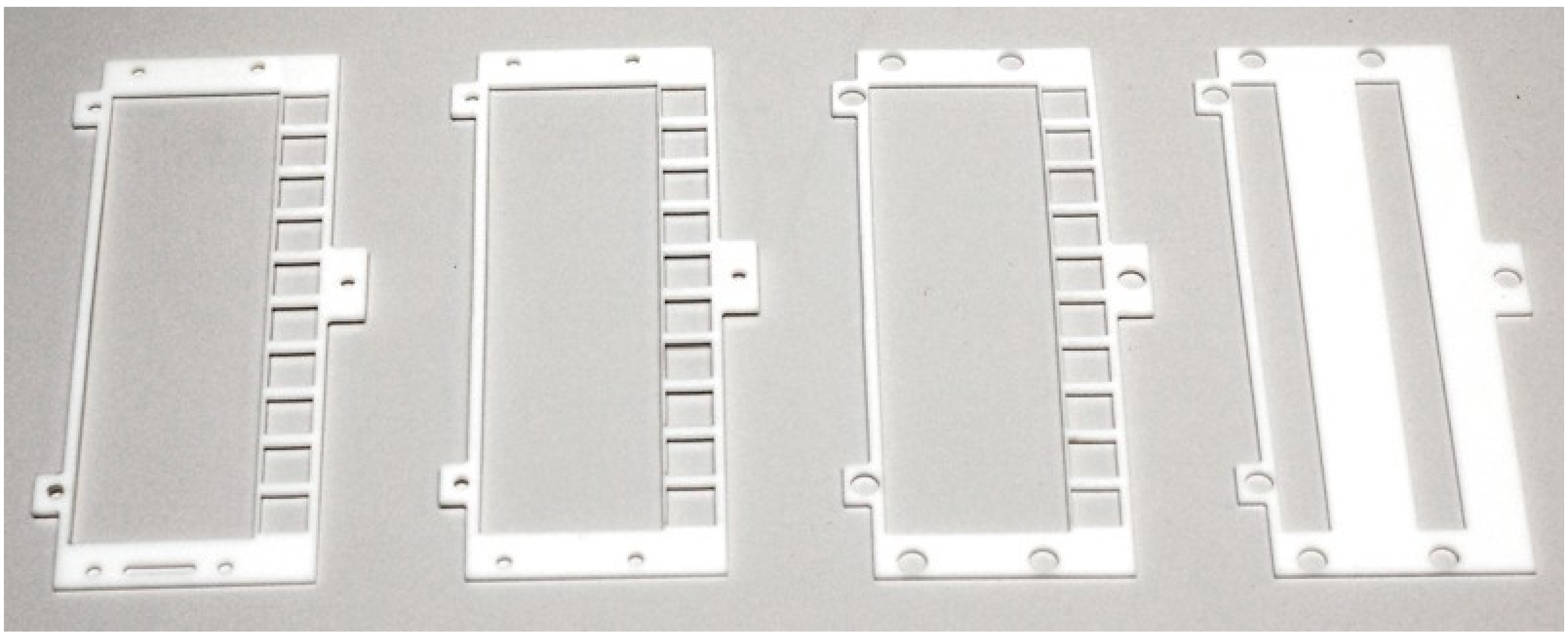
When installing the display board, you should pay special attention to the cases of the LED strips. They are installed on the board with no gap, so any unevenness of the case will not allow them to be installed evenly. The fact is that the pitch of the matrix segments is equal to 2.54 mm. 5 matrices with 10 segments each are installed in a row. A distance of exactly 2.54 mm is made between the holes for the outermost pins of neighboring matrices on the board, otherwise a “failure” of the segment pitch will be noticeable. If you measure the length of the matrix body, then in the upper part it is exactly 25.4 mm, but closer to the terminals the body expands a little. Probably, the compound inside, when solidified, slightly increases the volume. The expansion is insignificant,but in some cases the ends of the dies need to be lightly processed with a flat file.

Each line can be assembled from indicators of several colors. For example, you can set 30 green segments first, then 10 yellow ones, then 10 red ones. Colors can be freely selected. You can make the entire line in one color. If matrices of different glow colors are used, then in advance it is necessary to select limiting resistors for each type of matrix, so that visually the brightness of the glow is the same. This can be done using an external 3.3 - 5 V power supply. In my case, 270 Ohm resistors were needed for the green matrices, and 560 Ohm for the yellow and red ones. With other types of matrices, the denominations may be different.

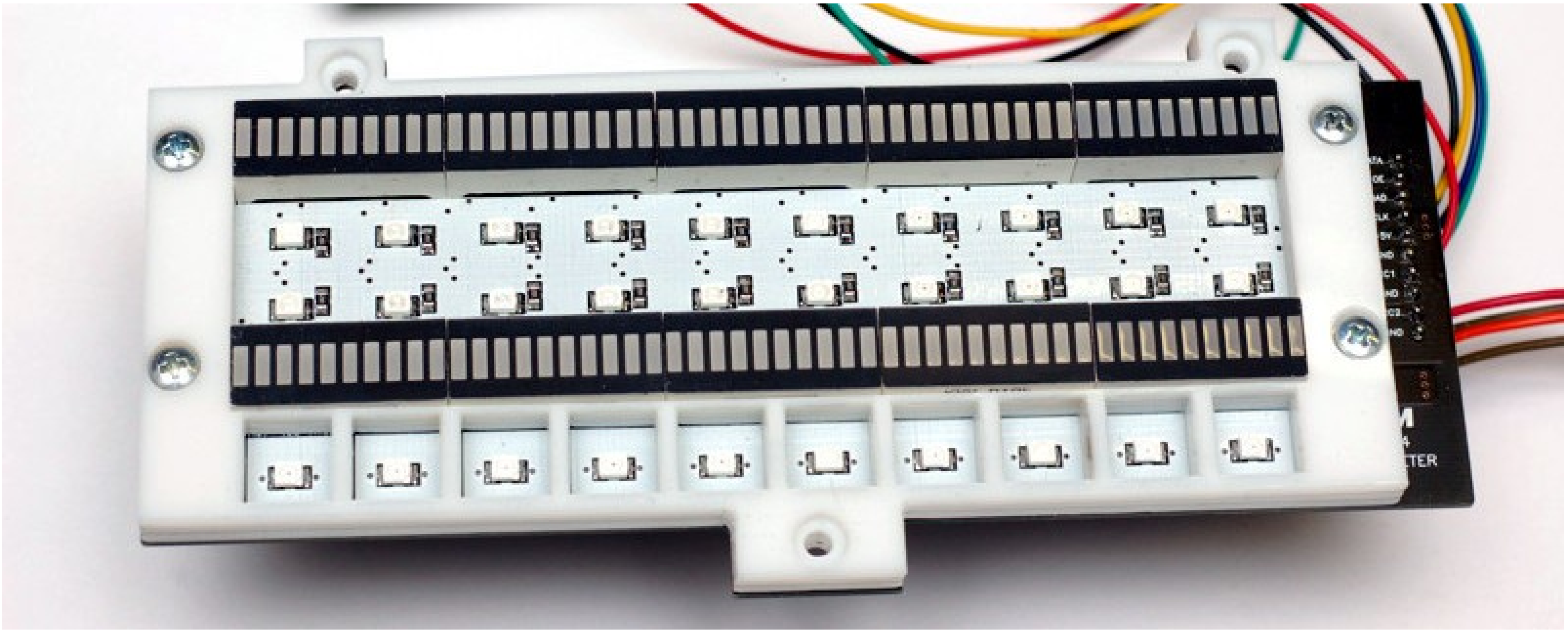
Scale backlighting LEDs, which are installed between the bars - any in the 3528 housing. Their color can also be chosen arbitrarily. I applied the same glow colors as the matrices. For these LEDs, it is also necessary to select limiting resistors in order to equalize the brightness. In my case, green LEDs needed 1k ohm resistors, yellow ones - 390 ohms, and red ones - 560 ohms. It is advisable to take LEDs from the same batch, otherwise their brightness may differ.

The situation is similar with the stencil backlight LEDs. Their colors can also be chosen arbitrarily, limiting resistors are needed the same as for backlight LEDs of the same color.

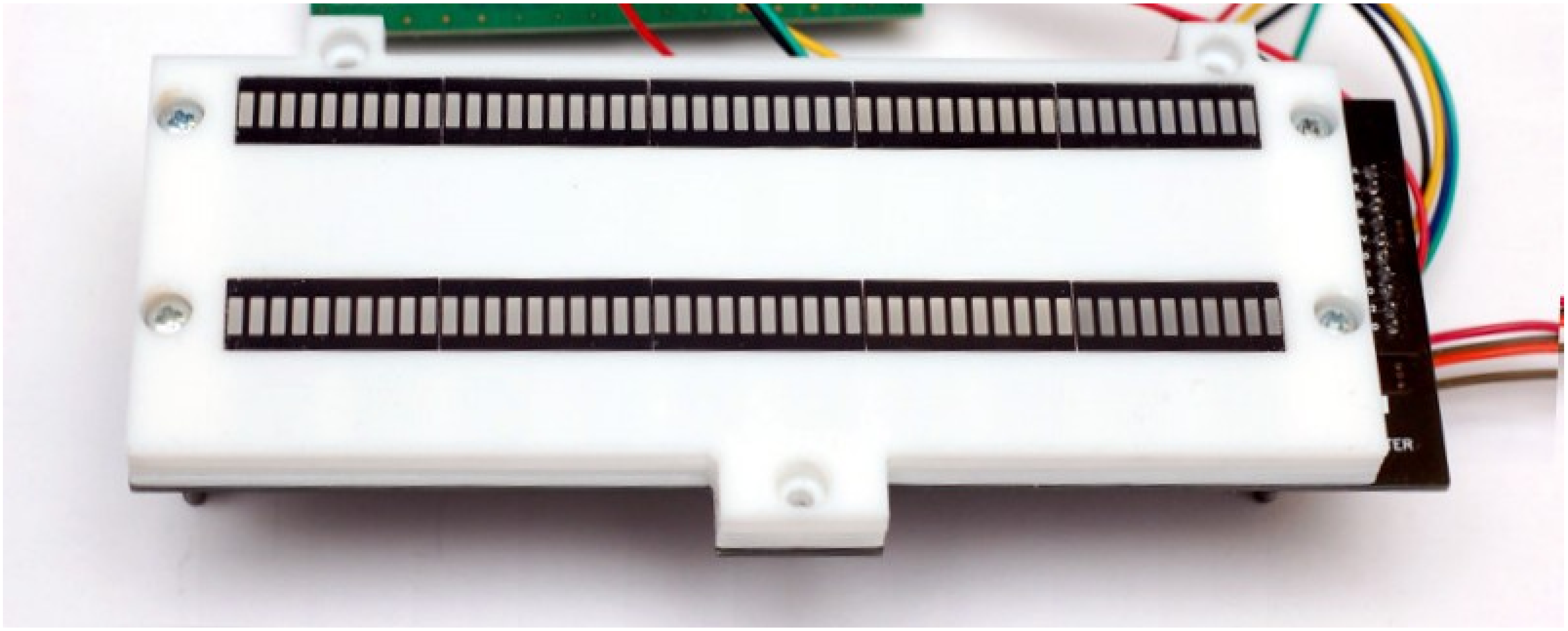
A diffuser in the form of a multilayer structure made of plastic sheet is installed on top of the display circuit board. One of the options is made of 2 mm thick milky plexiglass. With the help of laser cutting, 4 parts are made, which are then assembled in a pile and glued together. Part drawings can be downloaded at the bottom of the page.



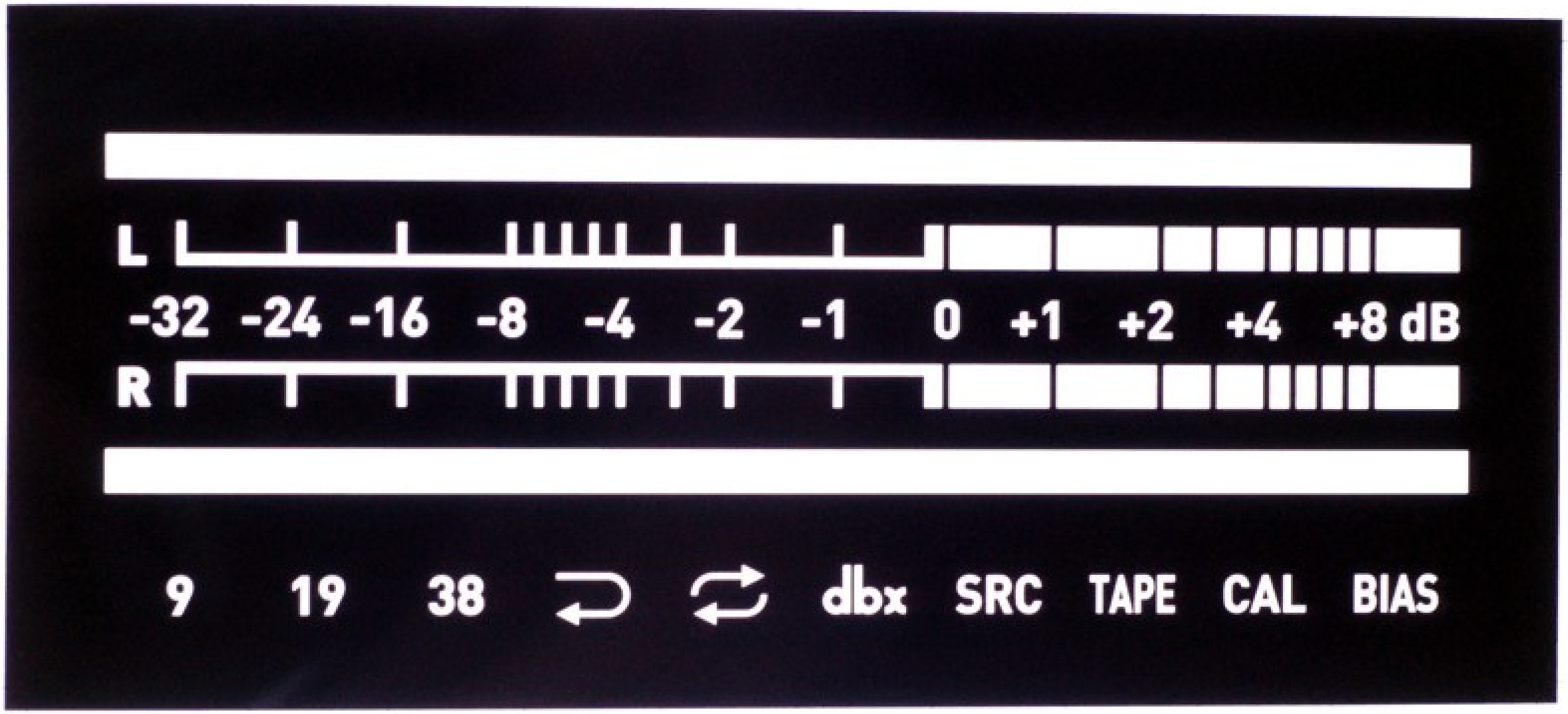
The three lower layers have through rectangular holes in the scale illumination area and for transparencies. The two lower layers are screwed to the board with screws, and in the two upper layers, the holes for the screws are made according to the size of their cap, which is recessed into them.



The topmost layer has slots for LED strips only. For backlight LEDs and transparencies, it acts as a diffuser.

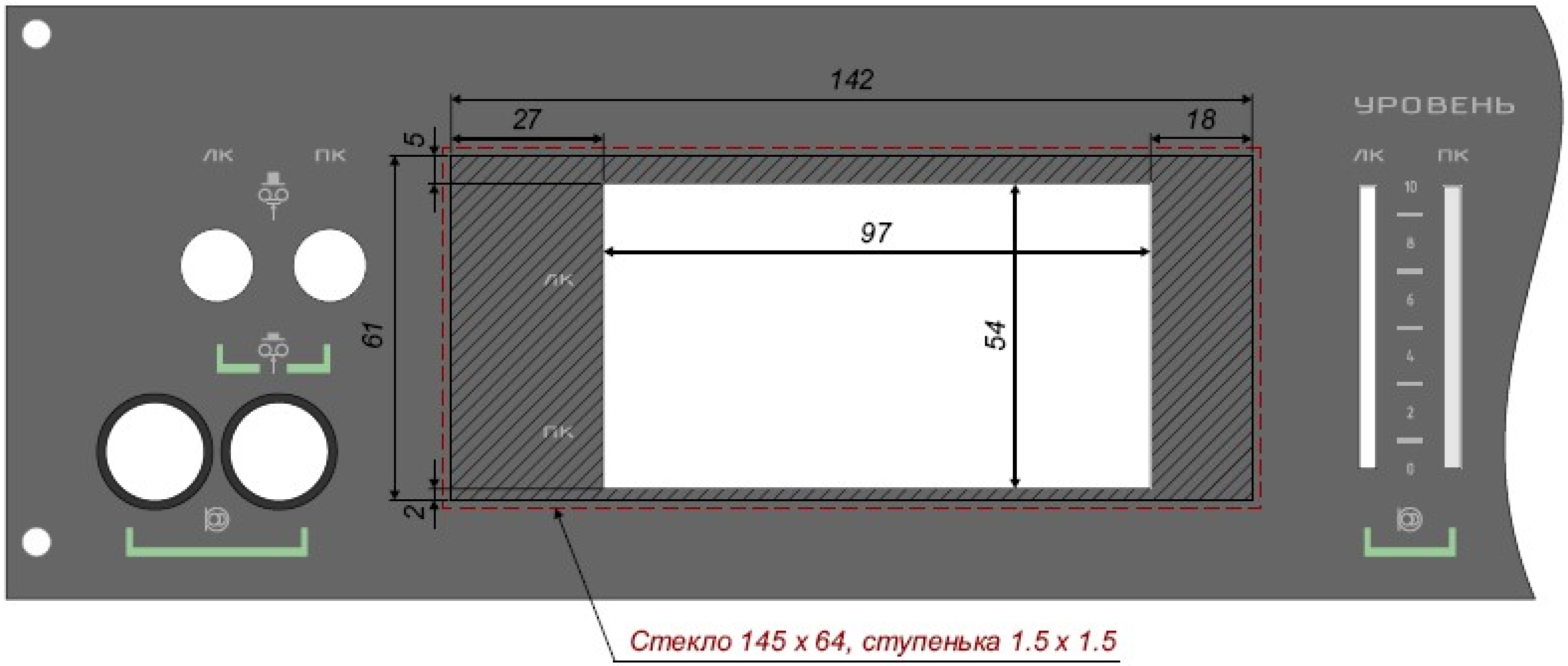


It remains to install the scale mask on top, securing it in the corners with a thin double-sided tape. The mask is made using a photo output. The mask drawing can be downloaded at the bottom of the page. The mask has transparent stripes for rulers, the height of which is slightly less than the height of the indicator segments. Thanks to this, the top and bottom edges of the rulers are perfectly flat, even if the matrices are not very evenly soldered into the board and "dance" in height. The stencil inscriptions have not yet been finalized. The part of the mask above the stencils can be replaced separately, it can even be printed on an inkjet printer. The meter with the mask installed is shown in the photo at the very beginning of this page.

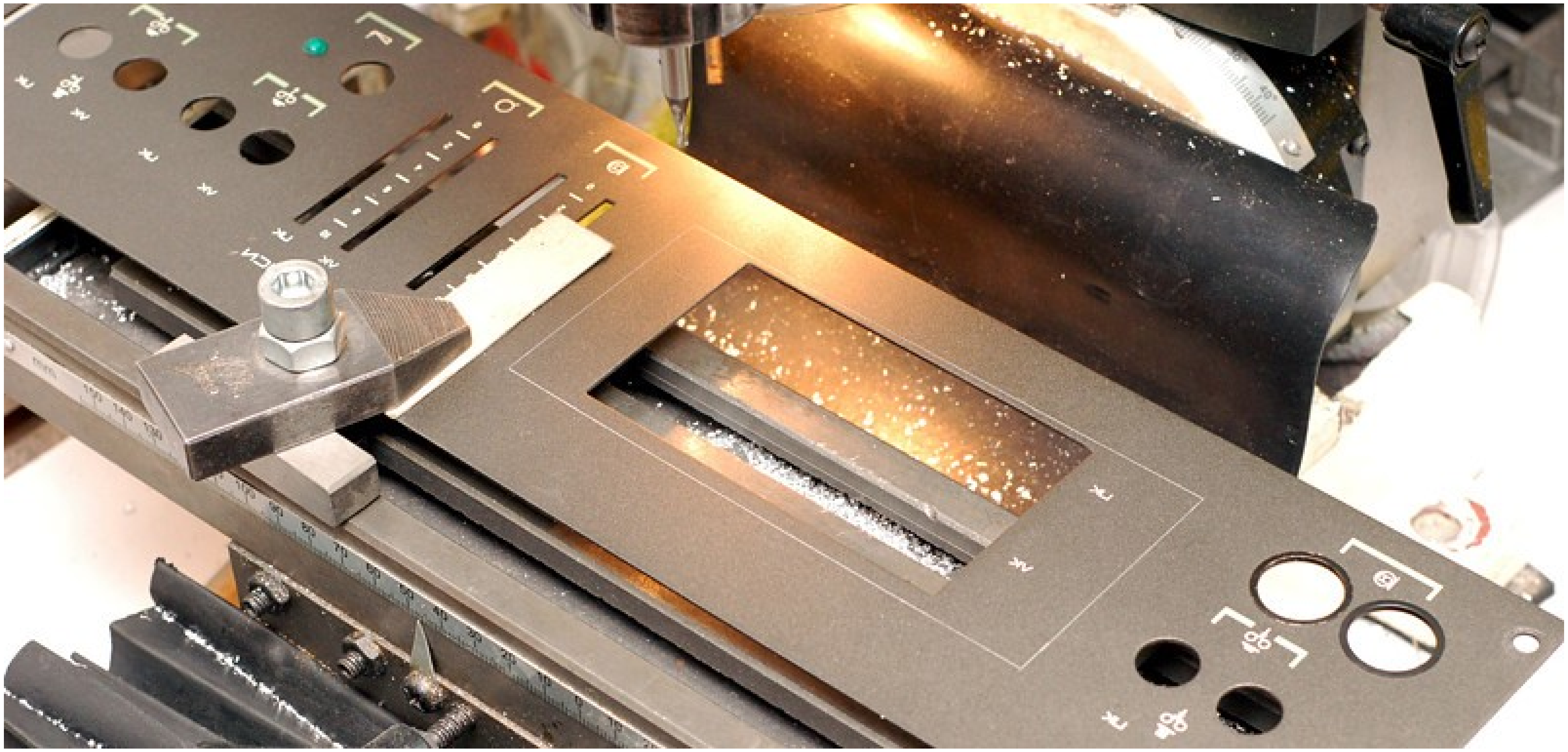


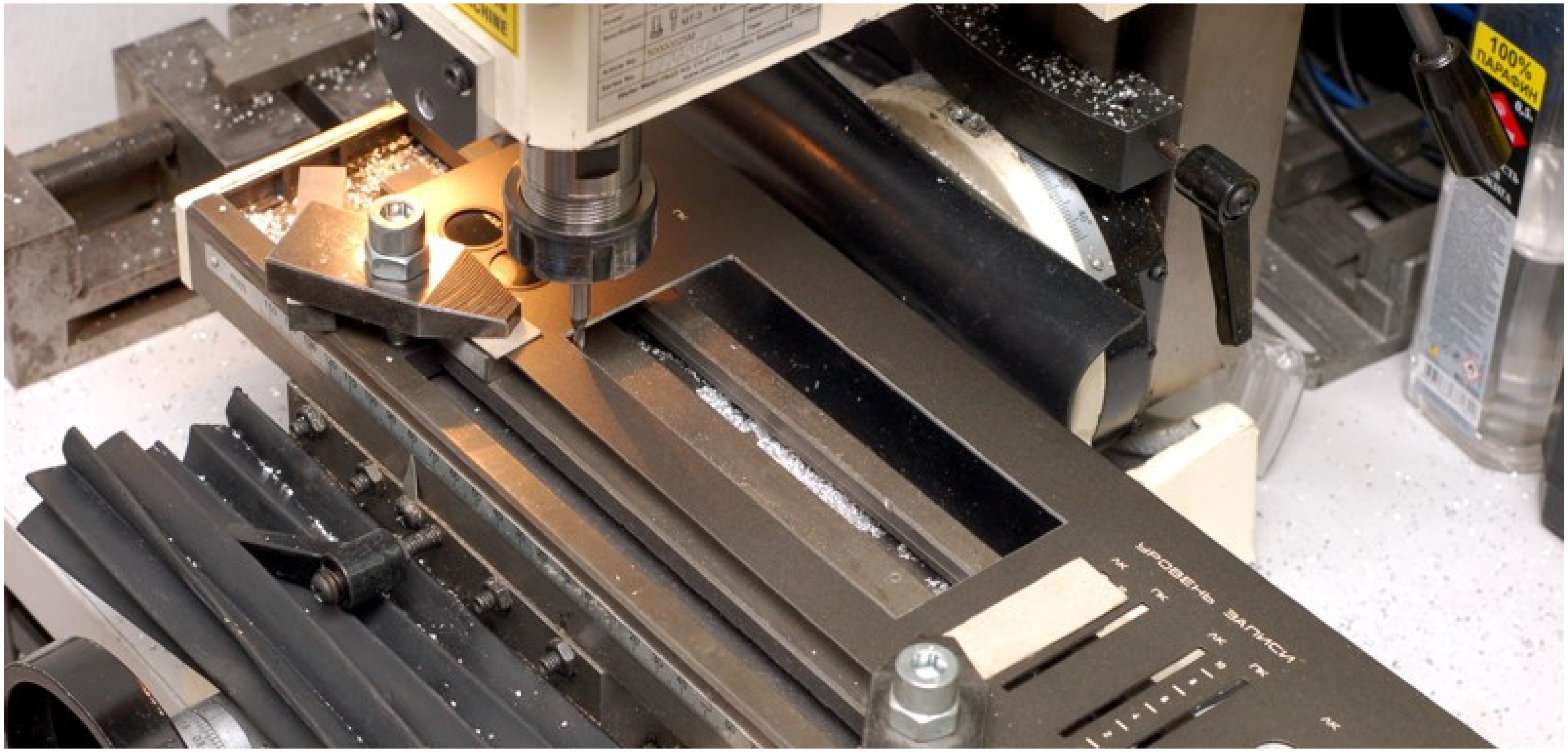
Alternatively, the mask can be laser engraved with Rowmark LaserLIGHTS S61 (black / white) film or similar. The white base of the film works well as a diffuser.

In the front panel of the tape recorder, to install a new level meter, it is necessary to increase the window size. A drawing of the new window is shown in the figure below.

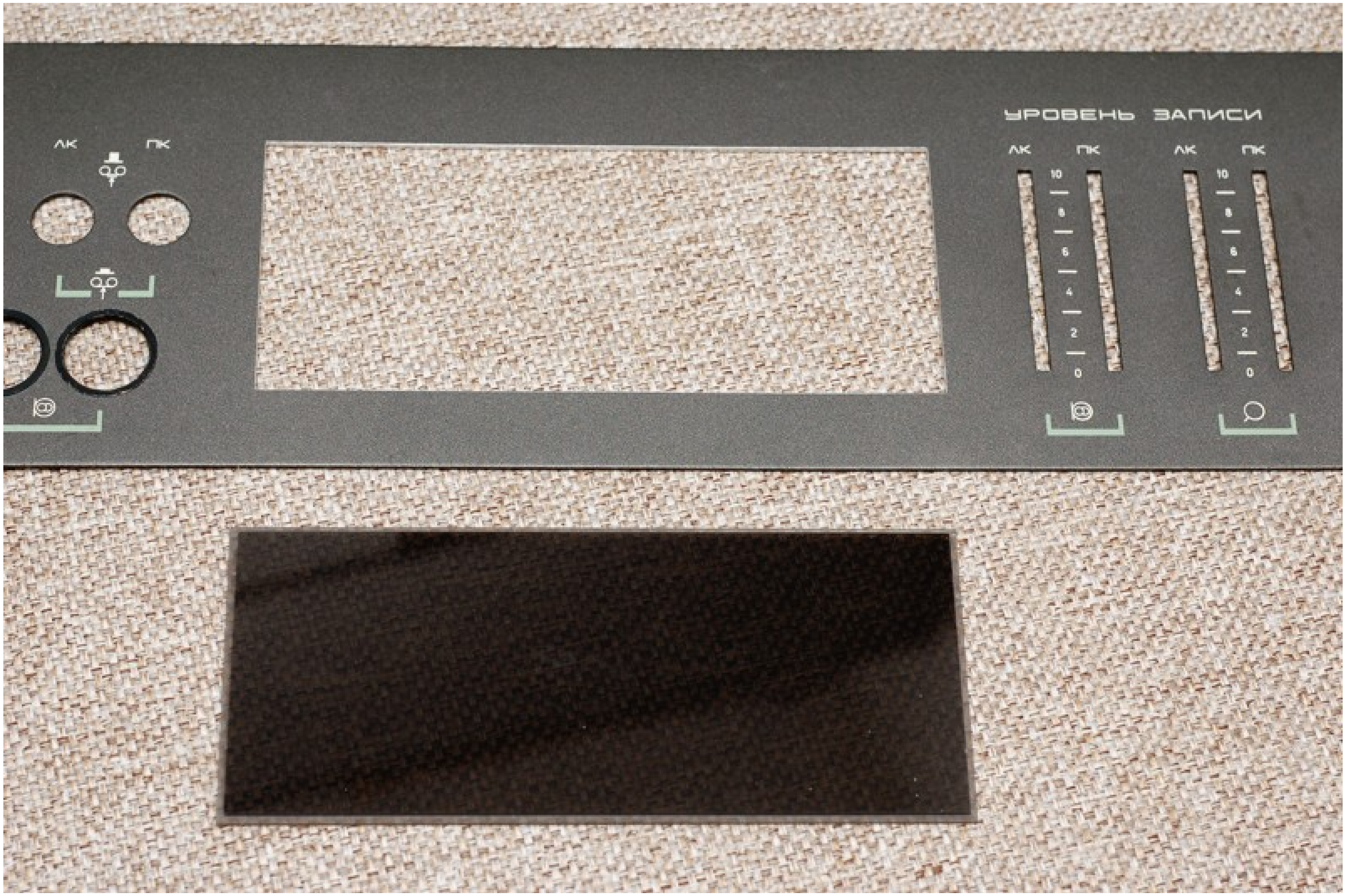


Cutting out a new window is easy. You can drill a number of holes along the contour, then combine them with a file, then align the contour with a wide flat file. With a milling machine, the work is greatly simplified.





The glass of the level meter is made of 3 mm thick smoky reinforced glass (in the extreme case - 2 mm). A step is milled along the contour to half the thickness of the glass. The dimensions of the glass are shown on the drawing of the new window. Alternatively, you can use clear glass and car tint film. If it is not possible to mill the step, the glass can be simply glued into the window. The only thing, in this case, the irregularities of the cut will not be closed.



To mount the new meter, standard studs and springs are used, only new nuts are needed. Since the nuts are recessed into the round recesses of the diffuser, they must be cylindrical and have a slot for a screwdriver. These nuts can be made from round M3 threaded posts. The photo below shows the fitting of the indication board without a diffuser.

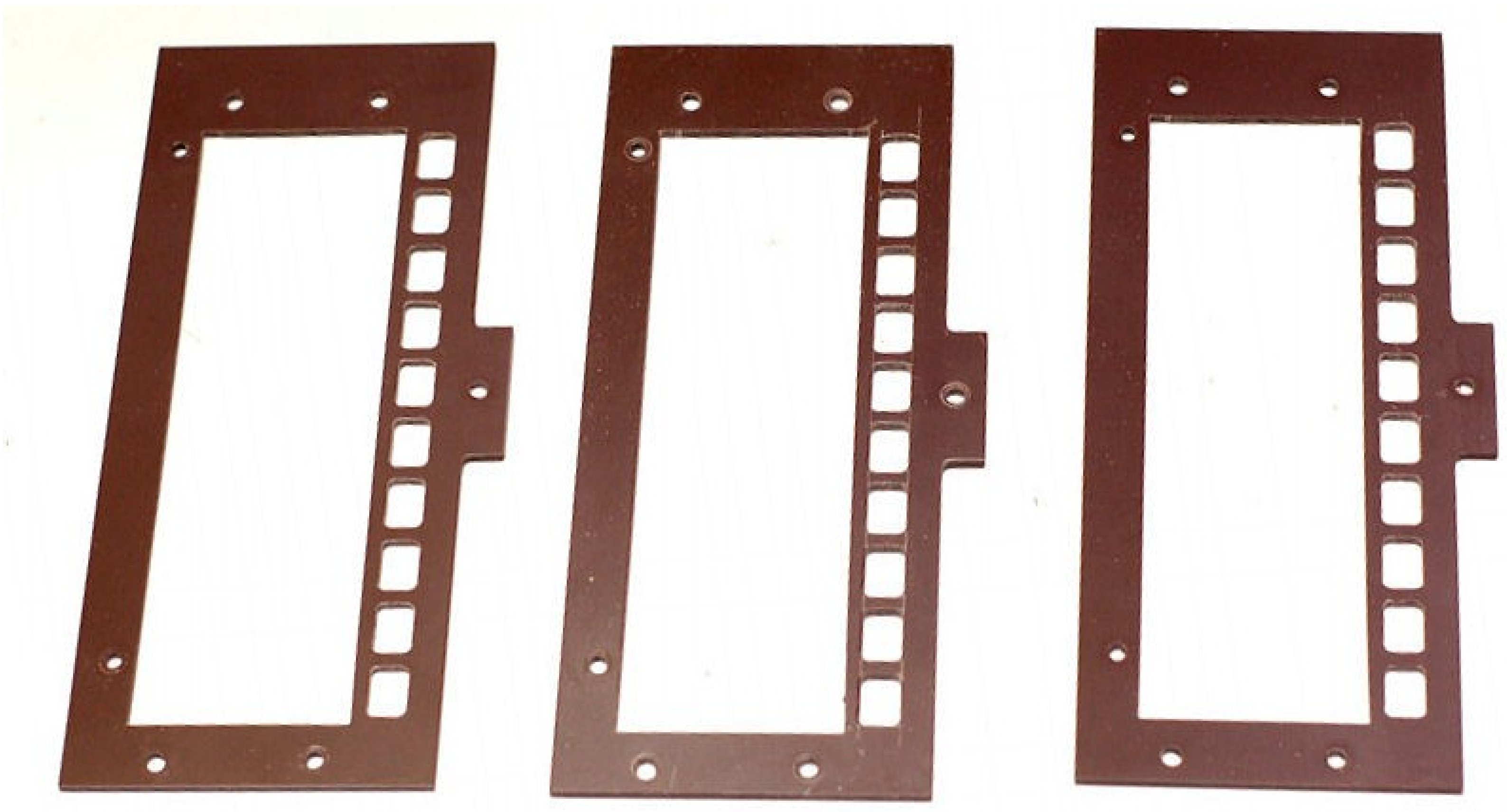
The revision of the front panel of the tape recorder is not approved by everyone. But in this case, the revision does not affect the inscriptions and does not change the style of the device. The new level meter window makes the recorder panel more attractive.

For comparison - a photo from about the same angle, but for a panel with an old level meter:

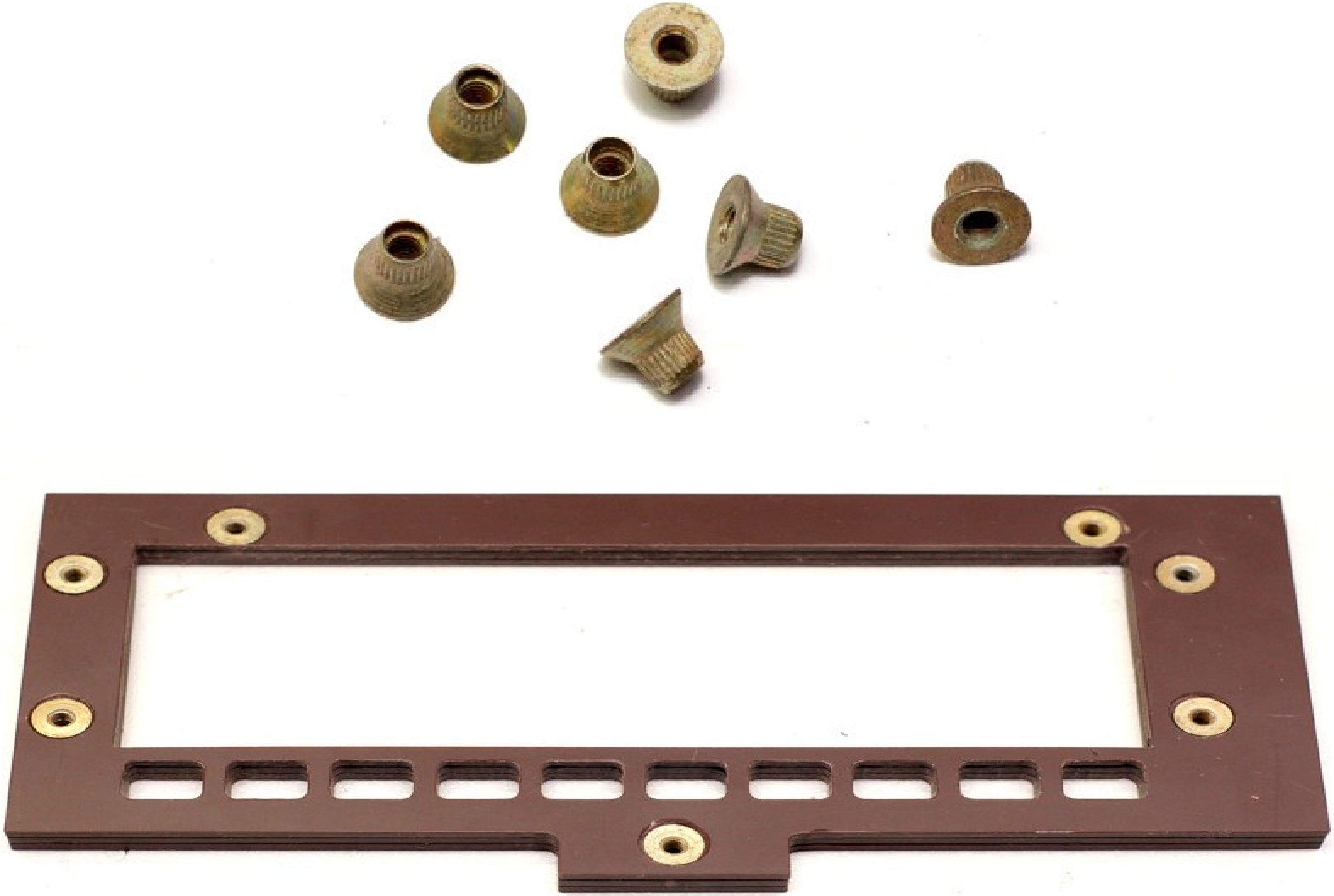
The diffuser design has several disadvantages. First of all, these are too thin bridges, which can be deformed during laser cutting. The size of the diffuser parts was originally made equal to the size of the display board. But this is not necessary, the window in the chassis of the tape recorder allows you to install a larger diffuser. Therefore, improved blueprints were developed. At the bottom of the page, you can download an archive with several versions of the drawings.

Another drawback of the diffuser is that light penetrates from the end of the glass, which is undesirable. It can be seen through the slots on the front panel near the connectors and controls. It is better to make the inner layers of the diffuser from an opaque material such as black acrylic. You can take a material of 2 mm (then there will be 3 layers) or 3 mm (then there will be 2 layers). Only the top layer is made of 2 mm thick milky plexiglass.

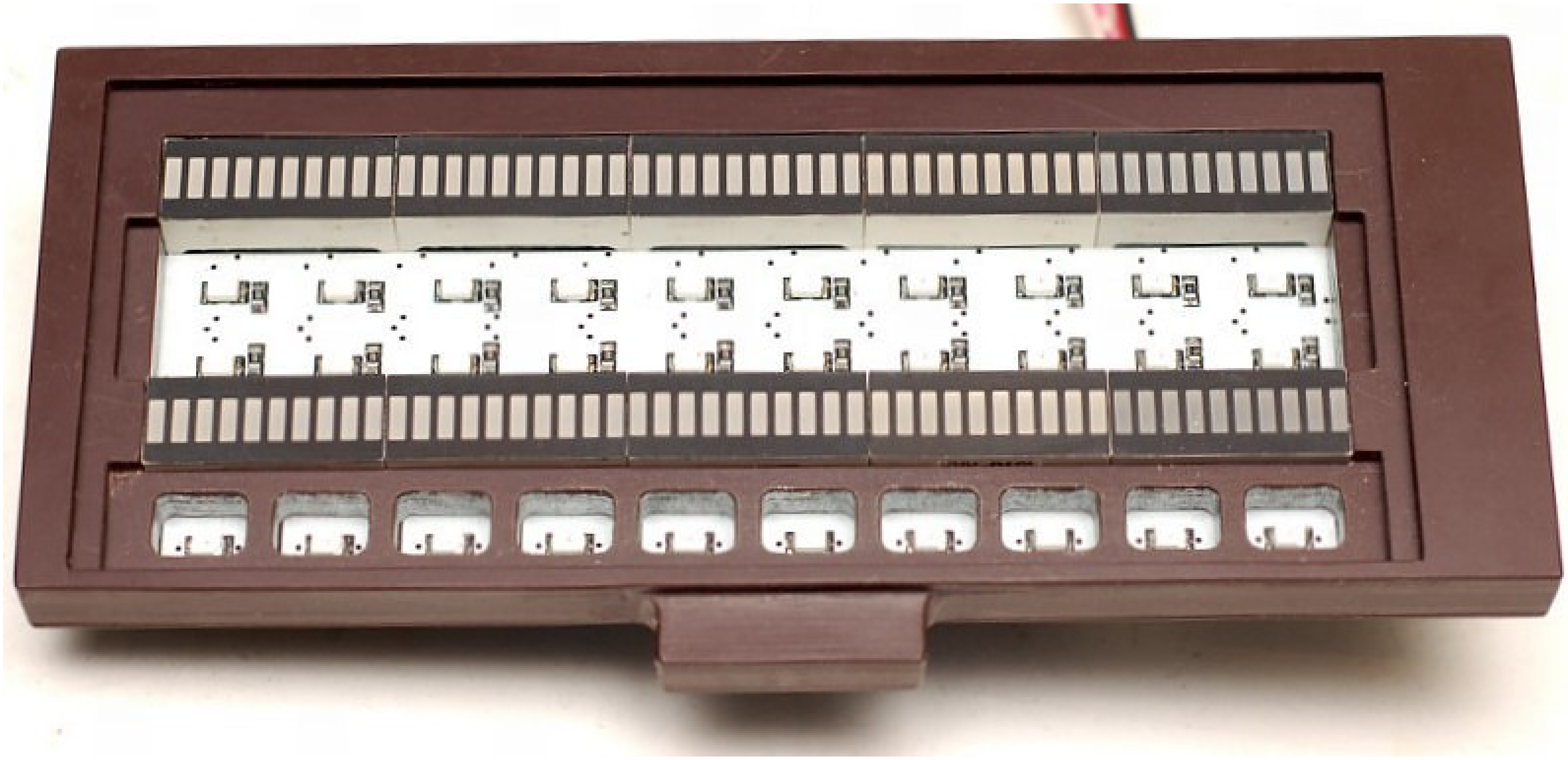
The end of the upper diffusing glass will still glow. To get away from this, diffusers can only be made above the scale illumination zone and above the stencils. Make everything else opaque. I also made this option on a milling machine. Although it is quite possible to do it using laser cutting. As a material, I took brown polystyrene 2 mm thick (black could not be found). The bottom three layers have cutouts for backlighting and stencils.

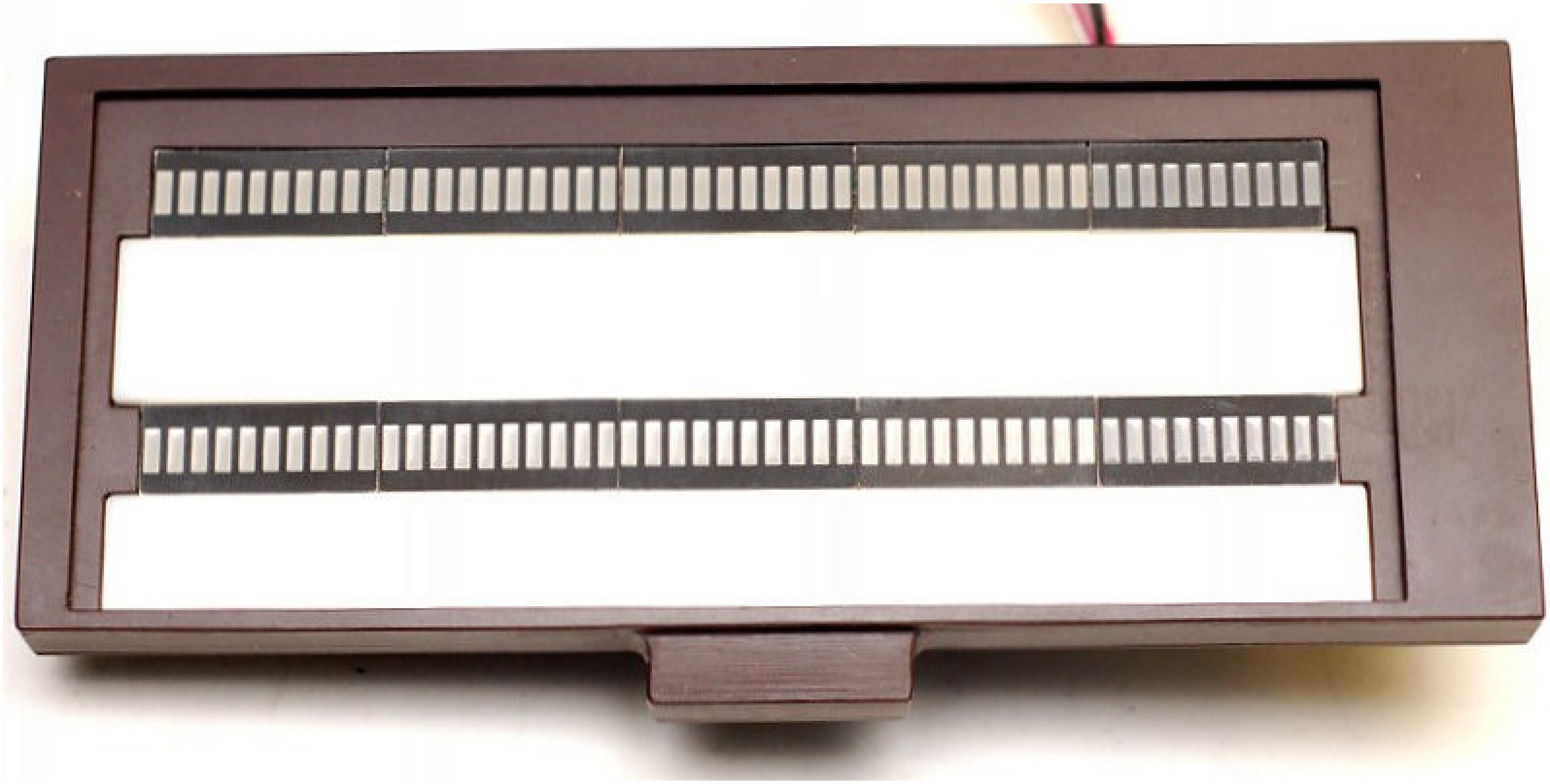


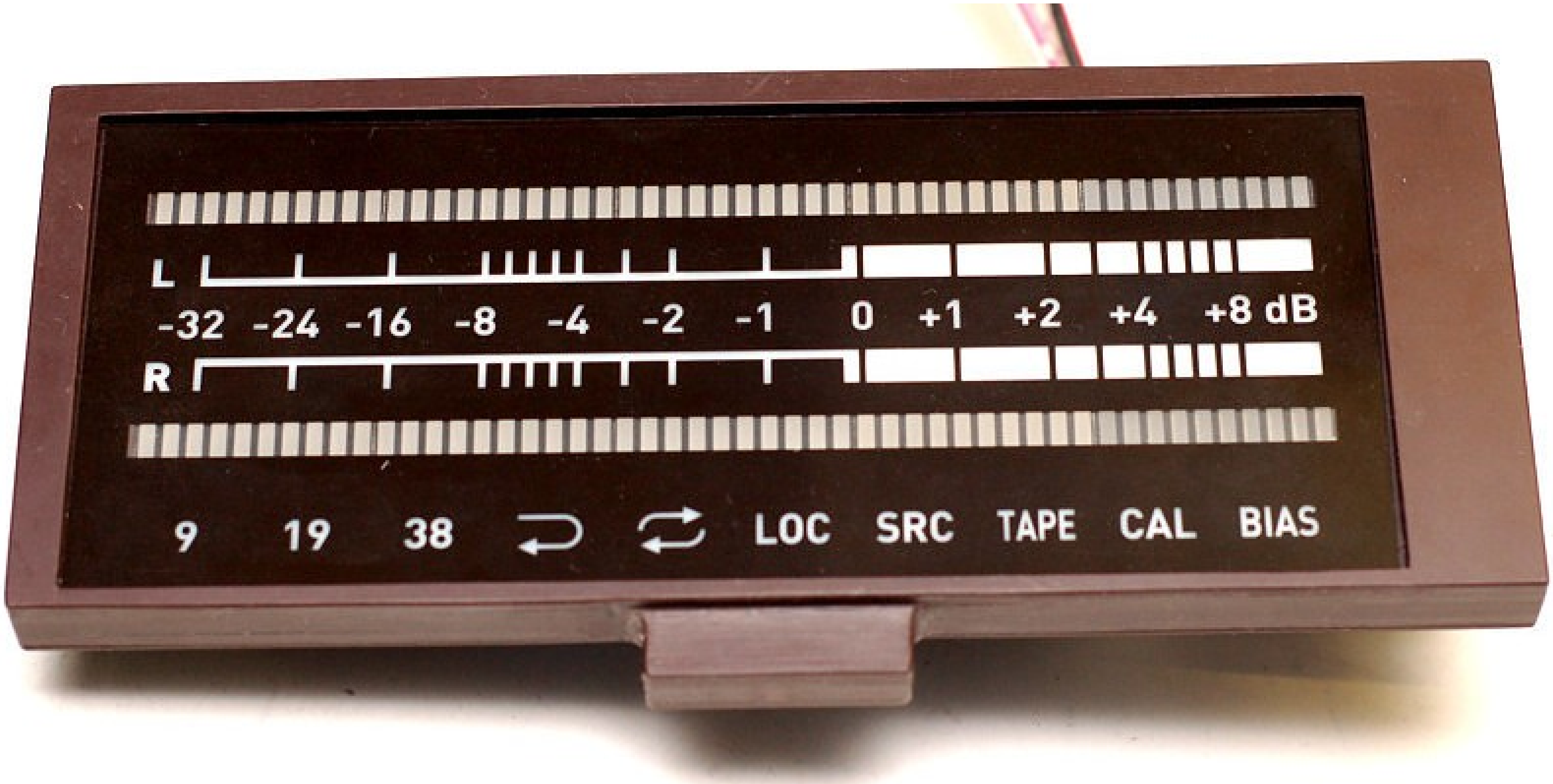
These three layers are glued together and nuts are pressed into them. You can do without nuts if you use self-tapping screws for assembly. Or screws can be glued in instead of nuts. In general, the design here can vary depending on desire and capabilities. In the lowest layer, grooves are cut in those places where the connectors are installed on the board.

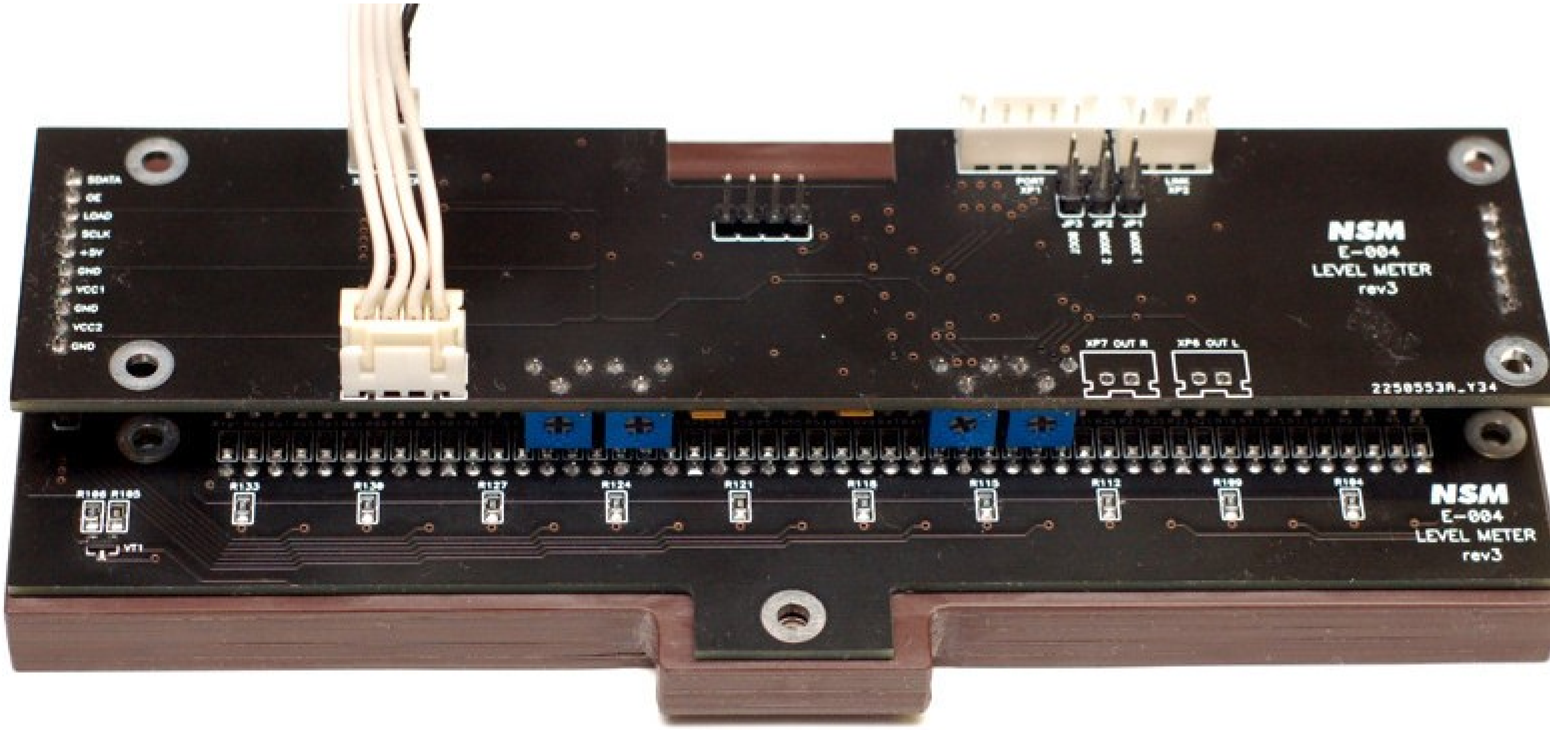


The fourth layer, which used to play the role of a diffuser, is also made of opaque plastic here. In it, seats are made for two milky plexiglass plates. Thus, the translucent diffusers will only be inside the opaque plastic structure, the light will not go out anywhere. Apart from the scale itself, of course. The walls of the stencil holes and the side walls of the scale illumination area can be painted with reflective silver paint. On top, you can make another layer in the form of a thin frame. The cutout is about 1 mm smaller than the window in the front panel. A scale film is embedded in this frame, and glass can be glued to the edges of the frame, which may not have a step.

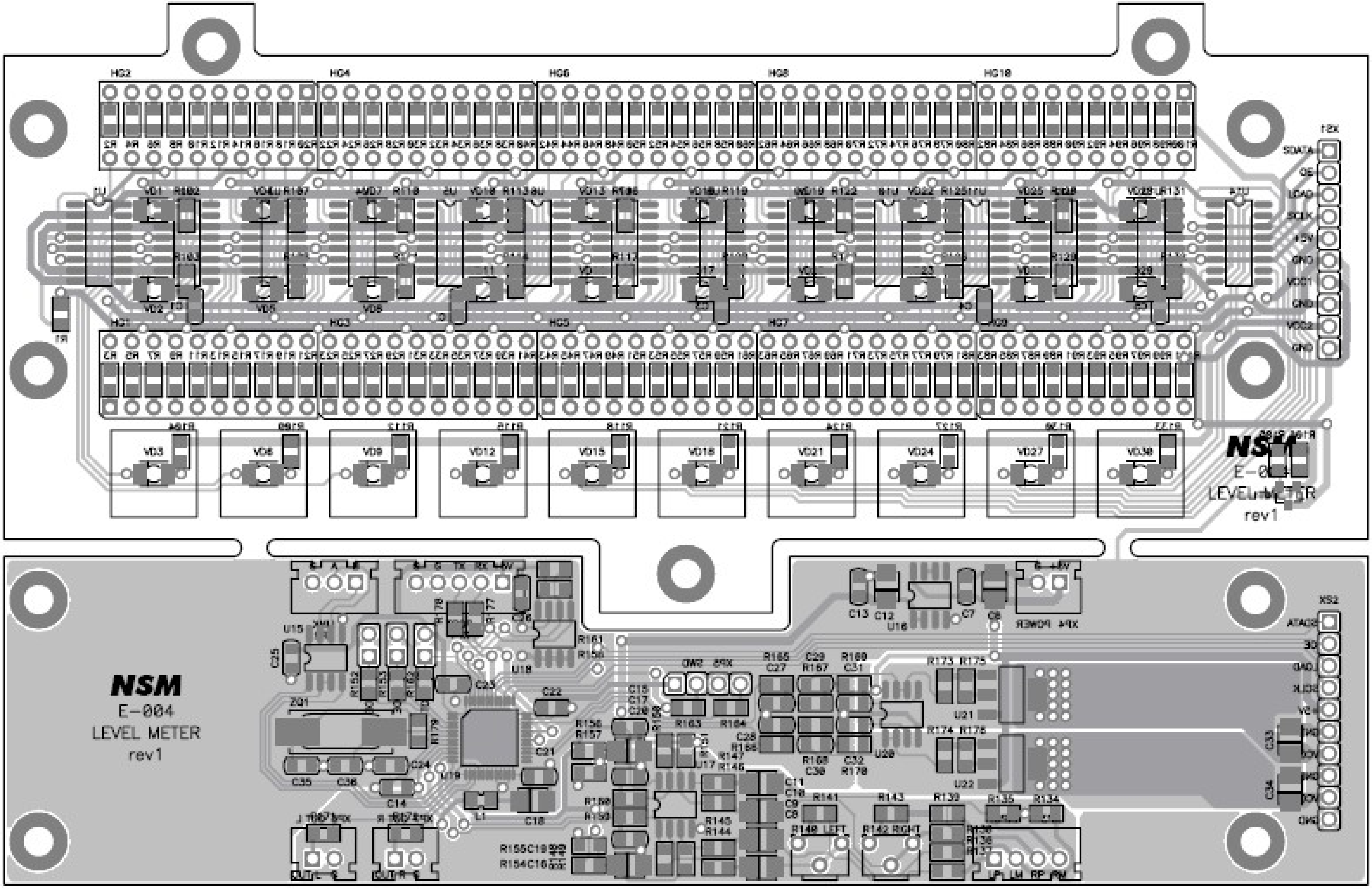


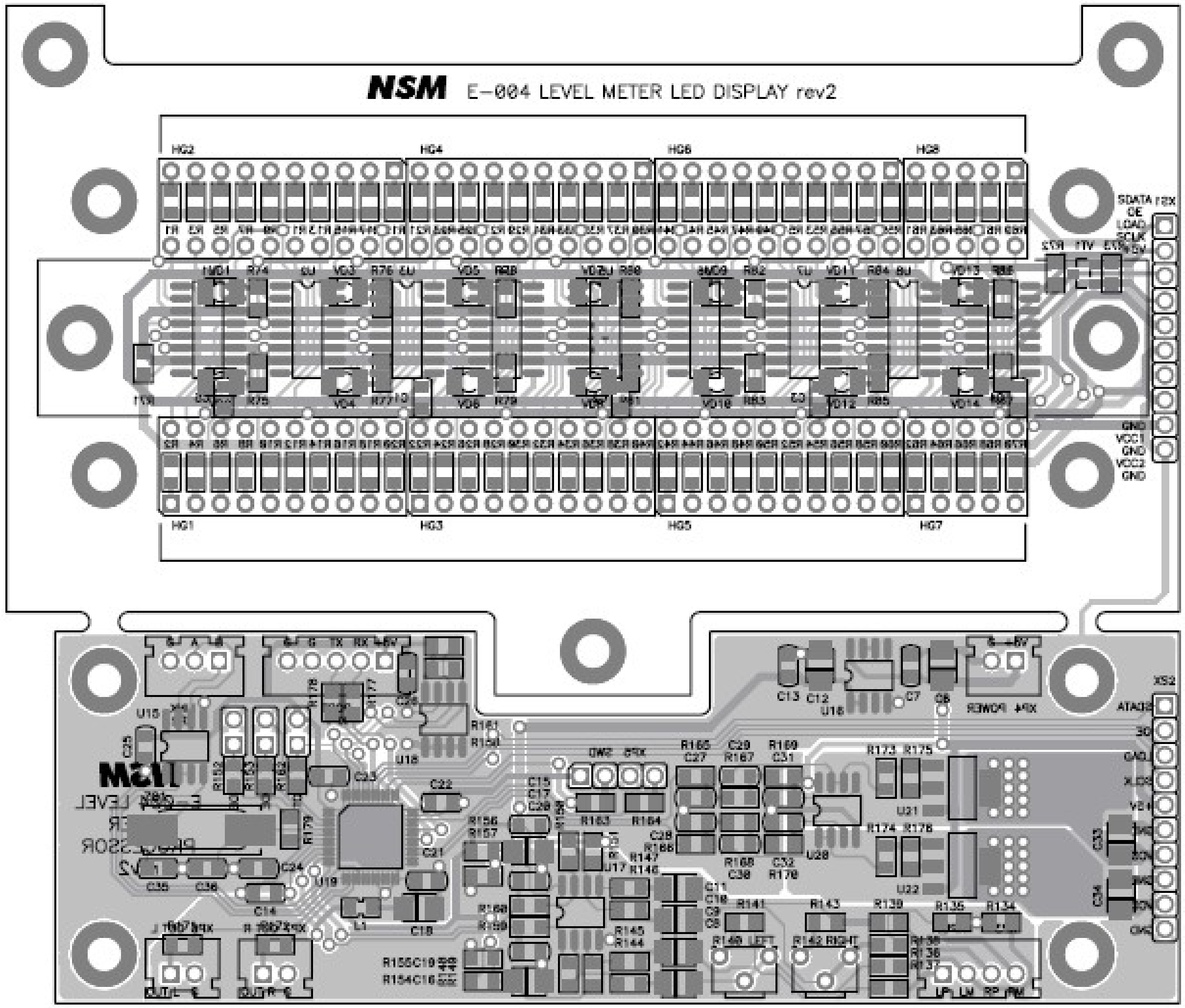



L
-32 -24 -16 -8 -4 -2 -1 0 +1 +2 +4 +8 dB
R
9 19 38 LOC SRC TAPE CAL BIAS



If you categorically do not want to cut the front panel of the tape recorder, then for such a case a "stripped-down" version of the printed circuit board with strips of 35 segments per channel has been developed. The firmware version for this version of the board has not yet been developed. The complete and stripped-down versions of the boards are shown below.


NSM
E-004
LEVEL METER
rev1
XS1
XS2
SDATA
OE
LOAD
SCLK
+5V
GND
XP4 POWER
LEFT
RIGHT

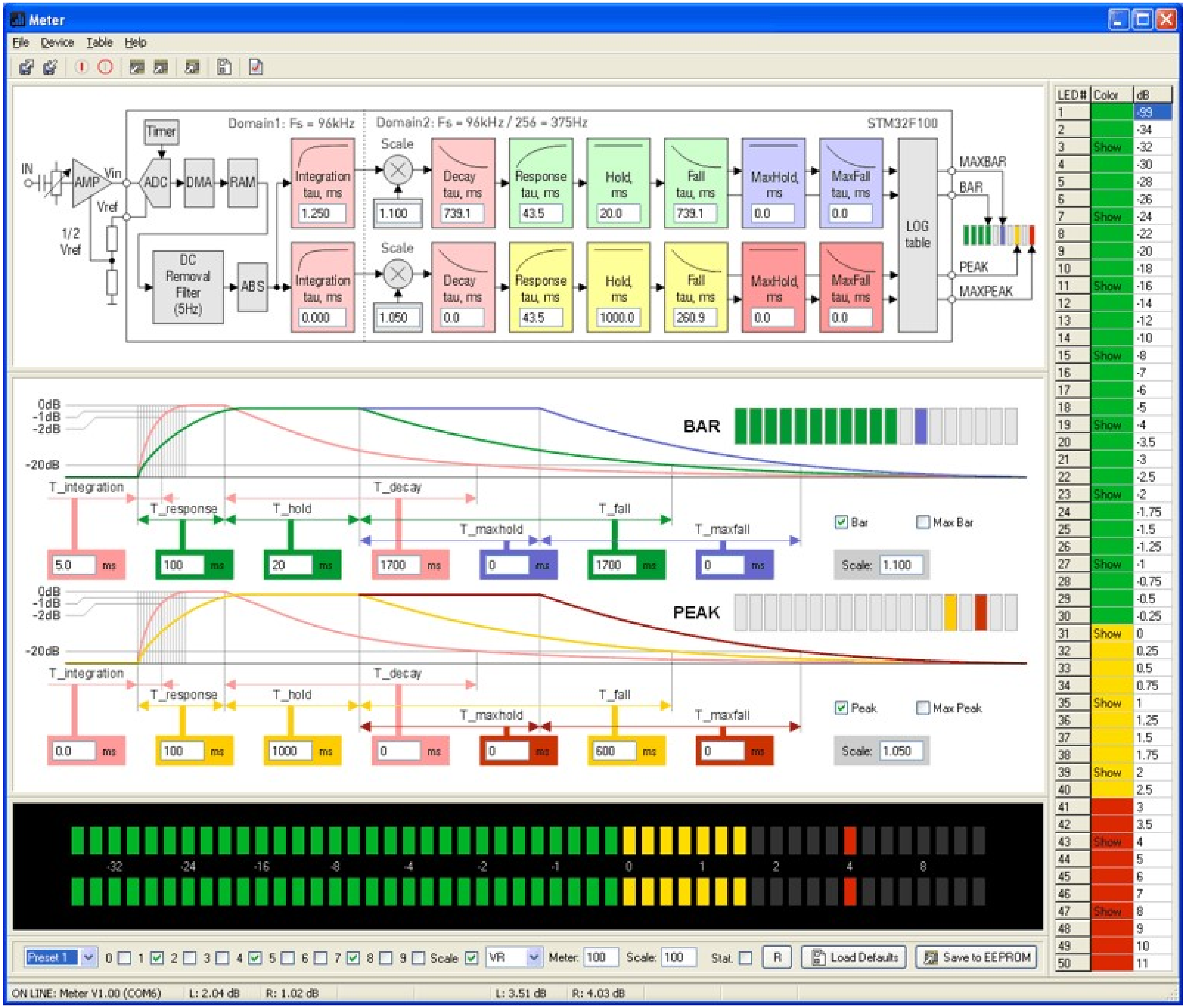


The appearance of a tape recorder with such an indicator, of course, will be worse.


ЭЛЕКТРОНИКА-004
-СТЕРЕО
ЭЛЕКТРОННОЕ УПРАВЛЕНИЕ
4 МАГНИТНЫЕ ГОЛОВКИ
РЕВЕРС
-95959
СДЕЛАНО В СССР
УРОВЕНЬ ЗАПИСИ
ГРОМКОСТЬ

Service program

The level meter, thanks to digital processing, allows you to set any values for the integration time, actuation, holding, retraction, and independently for the indication of the column and the luminous point. We can say that two independent meters work simultaneously here. In addition, the user can set any type of scale, defining a threshold in dB for each segment. And all this in 4 copies, which are loaded into the meter in the form of presets. It is difficult to manually set such a large number of parameters, therefore a special service program is required.



Meter
File Device Table Help
Domain1: Fs = 96kHz
Domain2: Fs = 96kHz / 256 = 375Hz
STM32F100
IN
AMP
Vin
Vref
1/2 Vref
Timer
ADC
DMA
RAM
DC Removal Filter (5Hz)
ABS
Integration tau, ms 1.250
Scale 1.100
Decay tau, ms 739.1
Response tau, ms 43.5
Hold, ms 20.0
Fall tau, ms 739.1
MaxHold, ms 0.0
MaxFall tau, ms 0.0
Integration tau, ms 0.000
Scale 1.050
Decay tau, ms 0.0
Response tau, ms 43.5
Hold, ms 1000.0
Fall tau, ms 260.9
MaxHold, ms 0.0
MaxFall tau, ms 0.0
LOG table
MAXBAR
BAR
PEAK
MAXPEAK
0dB
-1dB
-2dB
-20dB
BAR
T_integration
T_response
T_hold
T_decay
T_maxhold
T_fall
T_maxfall
5.0 ms
100 ms
20 ms
1700 ms
0 ms
1700 ms
0 ms
Bar
Max Bar
Scale: 1.100
0dB
-1dB
-2dB
-20dB
PEAK
T_integration
T_response
T_hold
T_decay
T_maxhold
T_fall
T_maxfall
0.0 ms
100 ms
1000 ms
0 ms
0 ms
600 ms
0 ms
Peak
Max Peak
Scale: 1.050
-32 -24 -16 -8 -4 -2 -1 0 1 2 4 8
Preset 1
0 1 2 3 4 5 6 7 8 9 Scale
VR
Meter: 100
Scale: 100
Stat.
R
Load Defaults
Save to EEPROM
ON LINE: Meter V1.00 (COM6)
L: 2.04 dB
R: 1.02 dB
L: 3.51 dB
R: 4.03 dB
LED# Color dB
1 -99
2 -34
3 Show -32
4 -30
5 -28
6 -26
7 Show -24
8 -22
9 -20
10 -18
11 Show -16
12 -14
13 -12
14 -10
15 Show -8
16 -7
17 -6
18 -5
19 Show -4
20 -3.5
21 -3
22 -2.5
23 Show -2
24 -1.75
25 -1.5
26 -1.25
27 Show -1
28 -0.75
29 -0.5
30 -0.25
31 Show 0
32 0.25
33 0.5
34 0.75
35 Show 1
36 1.25
37 1.5
38 1.75
39 Show 2
40 2.5
41 3
42 3.5
43 Show 4
44 5
45 6
46 7
47 Show 8
48 9
49 10
50 11

The main menu contains the following items:

File

- Open Preset ... - loading preset data from a file.
- Save Preset ... - writing preset data to a file.
- Exit - exit from the program.

Device

- Connect ... - establishes communication with the meter.
- Disconnect - disconnects the connection.
- Read Params - reading preset data from the meter.
- Write Params - writing preset data to the meter.
- Save to EEPROM - saves preset data to the EEPROM of the meter.
- Load Defaults - loading preset data by default (it does not save them to EEPROM).
- Settings - opens the settings window.

There are several options in the settings window. "Auto connect" - after starting or breaking the connection, the program will try to connect automatically. "Auto load parameters" - when the connection is established, the preset data will be automatically read by the program from the meter. "Update rate" - period of updating values in the status line.

Table

- Fill - filling the selected range of table cells with one value.
- Interpolate - interpolation of the selected range of table cells: the cells are filled with values with a constant step, which is calculated by the outermost selected cells.

There are also table menu items in the context menu, which is invoked by pressing the right mouse button on the table.

Help

- About ... - information about the program.

Most of the menu items are duplicated by buttons on the toolbar. All Buttons, like other program components, have tooltips.

The upper part shows a block diagram of the meter. Some blocks have input fields for parameters. The tau values of the filters are entered here.

Below is a graph that schematically shows the meter's response to a burst of an input signal. Here you enter the values for the integration time, actuation, falloff, retrace. These values are related to tau by simple relationships: the integration time is 4 times longer than tau, all other times - 2.3 times. This is due to the definitions of the standard mentioned above.

The Scale, Hold, MaxHold values on the block diagram and on the graphs are the same.

To enable display of rulers (Bar), maximum values of rulers (Max Bar), signal peaks (Peak), maximum values of peaks (Max Peak), there are check-boxes.

On the right is a table that allows you to set the scale. Like other parameters, the table is saved in a preset and for each preset it can be different. To edit a table, double-click on its line or press Enter. A window for editing a table row will open. Here you can set the color of the LEDs (it must match the color of the strips installed on the board), enable or disable the display of the level for this segment in the program, and also set the level value in dB. The same window is called through the Fill menu item. Before that, you can select with the mouse a whole range of table lines and set the necessary parameters for it. For example, if you only want to set the color for a group of rows, then for the rest of the parameters you can select "Not Changed".The color and sign of inclusion of the signature are not saved in the preset.It is stored in the general ini file of the program.

In the lower part of the window, there are rulers that are displayed in real time. This is a copy of the meter's scale. Below is the control panel.

Preset - allows you to select a preset number. There are preset selection jumpers on the meter board. If you rearrange them, the program will read the new preset and show the parameters for it. The same will happen if the preset is switched from the program. It should be noted that jumpers have priority, if you rearrange them or turn off / turn on the power of the meter, the preset will be set according to the jumpers. **The preset number is not saved in the EEPROM! When you turn on the power of the meter, the preset selected by the jumpers is activated.**

Checkboxes 0-9 and Scale include stencil LEDs and scale backlighting. Next, the brightness control method is selected: OFF - minimum brightness, VR - adjustment using trimming resistors, DAC - adjustment using a DAC (instead of trimming resistors, there should be jumpers, or the resistors should be at maximum), PWM - adjustment using PWM (this method interferes with the ADC, so not recommended). The brightness value is set separately for the rulers (Meter) and for the highlighting of the scale and stencils (Scale). All of these parameters are also stored in the preset.

The Stat checkbox allows you to turn on the display of statistics (the maximum level for the entire operation time of the meter). The statistics are reset with the "R" button.

Default - loads the default values for this preset. Duplicates the menu item of the same name. Does not store values in EEPROM.

Save to EEPROM - saves preset data to EEPROM. Duplicates the menu item of the same name. If, for example, you open a preset file, it will be loaded into the meter, but to save it to EEPROM, press this button.

The status bar on the left displays the communication status. When the port is not open, "OFF LINE" is displayed. When the device is connected, "ON LINE: Meter V1.0 (COM6)" is displayed (version number and port number, of course, may be different). If the port was open, but the device is not connected, then if the "Auto Connect" option is enabled, the program will constantly try to connect, while "CONNECTING ..." is displayed. If you need to work without a connection, then it is better to close the port, otherwise the interface may slow down.

To the right in the status line, the signal levels are displayed in dB in the form of numbers with a resolution of 0.01 dB. First, there are the values for the bars of the left and right channels (BarL and BarR), then their maximum values (MaxBarL and MaxBarR), then the peak values (PeakL and PeakR), then the maximum peak values (MaxPeakL and MaxPeakR). If the display of some value is disabled, then it is not displayed.

Default presets

Preset 1 - quasi-peak standard meter, integration time 5 ms, response time 100 ms, hold time 20 ms, retrace time 1700 ms. The point operates as a true peak meter, integration time 0 ms, response time 100 ms, hold time 1000 ms, retrace time 600 ms.

Preset 2 - quasi-peak standard meter, integration time 5 ms, response time 100 ms, hold time 20 ms, retrace time 1700 ms. The dot displays the maximum values of the bars, hold time 1000 ms, retrace time 600 ms.

Preset 3 - imitation of the standard level meter of the "Elektronika-004" tape recorder, integration tau of 12 ms (the time constant of the RC circuit is about 6 ms and a half-wave rectifier), the return tau is 1500 ms ... The point is not used.

Preset 4 - simulates a Dorrough 40-A2 meter, tau integration and 300ms rolloff. The point works as a peak (true peak) meter, integration time 0 ms, actuation tau 10 ms, hold time 25 ms, retrace tau 300 ms. Additionally, one more point is used to hold the peaks, the time is 2000 ms, there is no retrace.

Firmware version history

VERSION 1.00

+ First release. Communication with a computer is not supported. One preset with default parameters is loaded into the meter.
+ The bar operates as a standard quasi-peak meter with an integration time of 5 ms, a response time of 100 ms, a hold time of 20 ms, and a return time of 1700 ms.
+ The point operates as a true peak meter with an integration time of 0 ms, an actuation time of 100 ms, a hold time of 1000 ms, and a retrace time of 600 ms.
+ Brightness control - only with trimming resistors.

VERSION 1.01

+ Implemented communication with the computer.
+ Implemented presets.
+ Brightness adjustment in all possible ways.

VERSION 1.02 (service program)

+ Fixed a bug with displaying rulers when there is no constantly burning segment (with a level of -99 dB).
+ Added loading of the table into the device when editing it.
+ Added a preset to the Presets directory with an imitation of the operation of an arrow meter.

VERSION 2.00

+ Added support for 35 and 50 segment meter options. The firmware files are located in the Meter35 and Meter50 folders. If the external EEPROM contained some data, then after flashing it is recommended to press the "Load Defaults" and "Save to EEPROM" buttons in the service program for each of the presets. This will eliminate the influence of "garbage" in the EEPROM on the operation of the meter.
+ Fixed a bug with adjusting the brightness using the DAC.
+ In the service program in the "Settings" (F9) menu there is a choice of "50 Segments" or "35 Segments". This changes the number of lines in the table and the number of segments in the scale.
+ Changed the logic of the table context menu, now when selecting several lines, the option "Not Changed" is offered for parameters whose values in the selected lines are different. This protects against accidental erasure of data within the allocated range of table rows.

VERSION 2.01 (firmware), 2.10 (service program)

+ Added support for 40 segment meter option.
+ Added "40 Segments" item to "Settings" (F9) menu.
+ Pin2 (PC13) for versions 50, 35 is not used, for 40 - turns on the backlight.

VERSION 2.11 (service program)

+ Fixed a bug with saving the state of LEDs (stencils) to the .pre file.

VERSION 2.02 (firmware), 2.12 (service program)

+ Added check for writing to EEPROM. If an error occurs, the service program displays the message: "Command EESave failed: EEPROM error."

Links:

1. AV Nikonov. Audio signal level meters. "Radio and Communication", 1981.

2. Demo video on Youtube

3. Demo video of the service program on Youtube

Downloads:

meter50_sch.pdf (354 kB) - schematic diagram (50 elements).
meter35_sch.pdf (265 kB) - schematic diagram (35 elements).
meter50_pcb.pdf (471 kB) - printed circuit board (50 elements).
meter35_pcb.pdf (341 kB) - printed circuit board (35 elements).
meter_scale.pdf (16 kB) - drawing of the indicator scale.
meter_corel.zip (1353 kB) - drawing of the diffuser and scale (Corel Draw 11, pdf).

- meter_pcad.zip (1069 kB) - PCB layout files (PCAD 2006, Gerber, Bom).
- meter_source.zip (798 kB) - firmware, source code (C ++ IAR EWARM, hex).
- meter_soft.zip (946 kB) - utility program (C ++ Builder 6, exe).

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