The Pocket "Color Organ"[1]

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Introduction:

The Pocket Color Organ is a relatively simple and entertaining electronics project to build, making it a good choice for beginners and experts alike. It only requires a handful of common parts and yet it exposes the builder to some very useful fundamental circuit elements and concepts – not to mention putting on a fairly cool light show when completed.

The heart of the Color Organ is the LM324 Quad "Op Amp" (operational amplifier) IC. This chip is a very versatile and thus a very common component used in a variety of consumer products (e.g. radios, graphic equalizers, etc.). The Color Organ circuit uses the individual amplifiers in the LM324 to form three distinct audio filter sections, which separate the input audio into three specific bands: Low range (around ~75 "cycles per second", or "Hertz"), Mid range (~400 Hz), and High range (~1100 Hz). The output from each section is amplified slightly and then used to drive three separate LED (Light Emitting Diode) strings.

Note that the LM324 can be configured in either a dual supply mode (e.g. +6v on pin 4 and -6v on pin 11), or in a single supply configuration (e.g. 12v on pin 4 and ground on pin 11). Since we are using this latter option, we must shift the signal DC level up a few volts or we will chop off the bottom alternations of our audio signal (we accomplish this DC biasing using the voltage divider resistors shown on all of the "+" amp inputs, i.e. pins 3,5,10,12).

Construction:

This document assumes the reader has some basic experience with electronic parts and can (for example) read a resistor color code (if not, we suggest reading some introductory material first, such as a Boy Scout electronics merit badge handbook or NAVPERS training manual (etc.); these handbooks can be obtained by contacting your local Boy Scout office or through inter-library loan). Before beginning construction, first accumulate and inventory all the required parts. This will make it much easier to complete the project once you start, plus it will get you familiar with each of the components used in this circuit.

As you assemble the project, do so *one section at a time.* Place the components on the board in the layout shown in Figure 1, keeping all the parts as well separated as possible to make it easier to troubleshoot later. Building and then testing *only one section at a time* will help isolate problems to a single section before proceeding (see below).

To avoid damaging the LM324 when soldering, and to make troubleshooting / replacement of the chip *much* less *painful* down the road, we *strongly* recommended using a 14 pin socket to hold the chip; do NOT solder directly to the chip. Please note that the chip and socket have an index notch to mark pin 1. Mount the socket and later (*after* all soldering is done) the chip *exactly* as shown in the figure. *If you put the chip in upside down and apply power, the chip will most probably be destroyed*. Before inserting the chip, double check that 9v DC is present on pin 4, and ground is connected to pin 11.

A common practice in building and troubleshooting all complex projects, is to divide the overall project into functional sections. In this project, the circuit can be broken down into *five* distinct sections: the Power Supply, Audio Input/pre-amp, Low range filter (L), Mid range filter

1 See "EpiphanyBySteveLee.com" for additional information to be added over time.

(M), and the High range filter (H) sections. We therefore recommend building this project by sections, starting with the easiest section (i.e. the Power Supply), followed by the Audio Input/pre-amp section, etc, testing and debugging each section *before* proceeding to work on the next. A Digital Volt Meter (DVM) should suffice for testing, or better yet an Oscilloscope (if you have one).

As for the placement of the LED's themselves, you may want to randomize the location of each LED a little (rather than place them in a straight line), for visual effect. Just make sure they are wired with their cathodes (black line side of the diode) towards ground, and that wires from one string do not short to any other string. If you accidentally flip one diode around, that whole string will not light up when voltage is applied. It would be a good idea at this point to verify that each LED is good (e.g. by applying 9 volts through a 470 ohm resistor to each LED), and then mark the cathode side with a permanent marker to minimize errors.

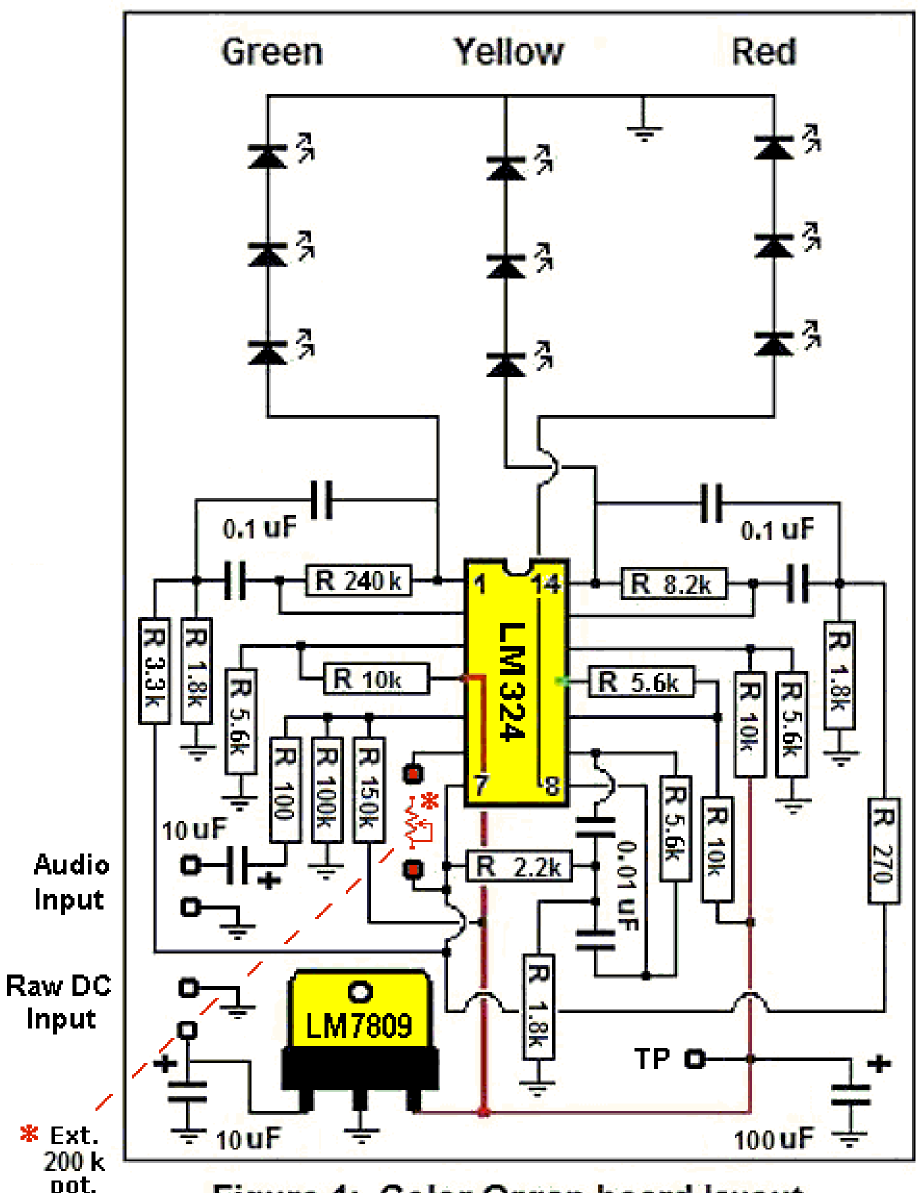
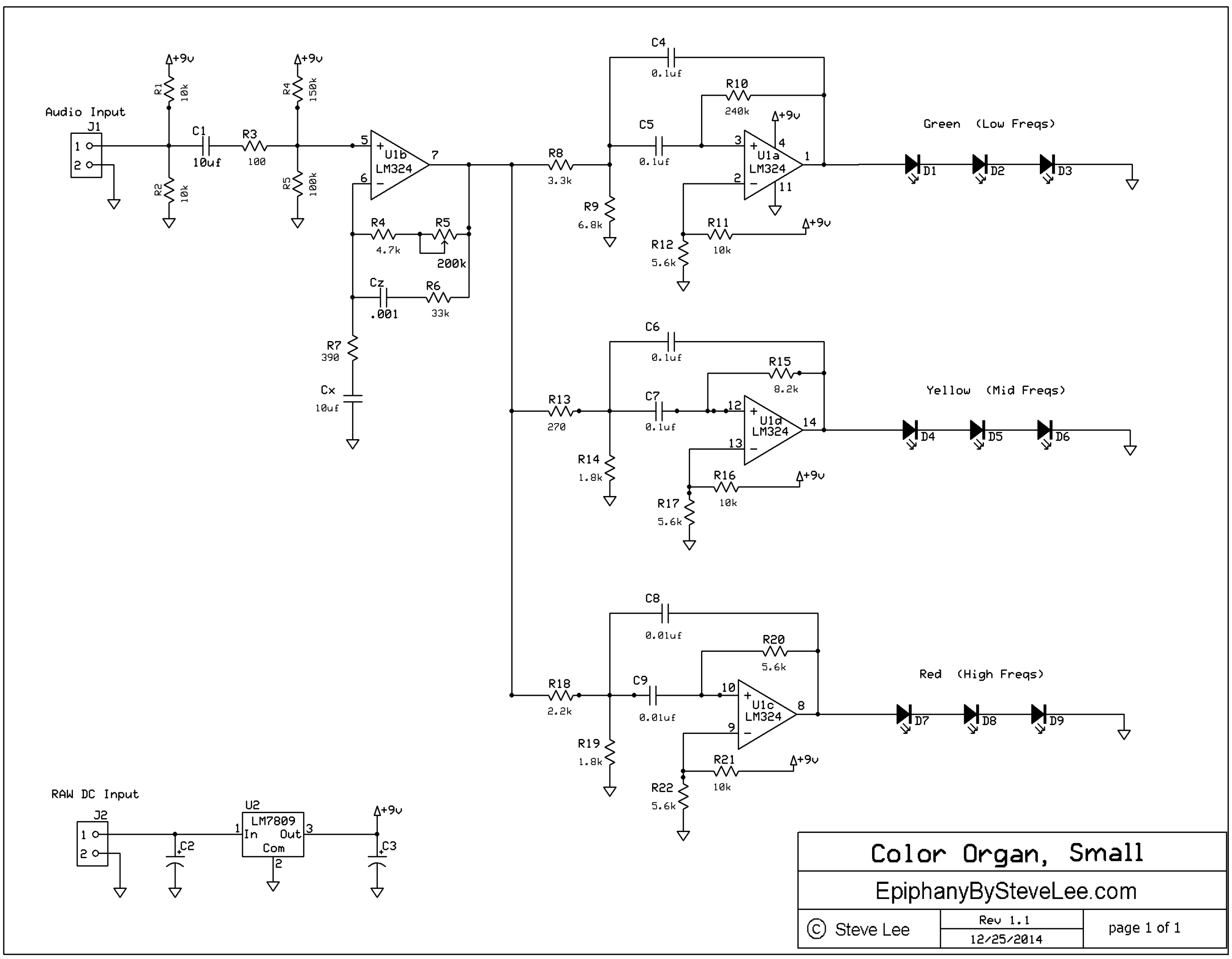


Figure 1: Color Organ board layout.

Power Supply: The primary component in the Power Supply is the LM7809 chip. The LM78xx family of power chips are cheap, very easy to use, highly effective, and thus widely used in a large number of consumer electronic devices on the market. An LM78xx takes a rough DC input on its left-most pin (when viewed from the front) – for example from a “wall wart” power transformer – and produces a smooth DC voltage on its output (its right most pin) at the specified value (9 volts in this project; the raw DC input used should be ~3v higher, or in this case roughly 12v DC).

Audio Input Section: Since the LED sections use only three of the four Op Amps in the LM324, we will use the unused amp on pins 5, 6, and 7 to buffer the input audio. This will allow us to easily match the impedance of the audio source you choose to use (e.g. pocket radio, condenser microphone, MP3 player, etc.), as well as help minimize any “loading” affects this circuit might have on your audio input device.

Filter Sections: All three audio filters use the same basic design, making the component placements around each of the three filters very similar. Each filter is fed audio from the output of the Audio pre-amp (pin 7) via one input resistor per filter. The amplified audio output signal from each filter is then used to drive each of the three LED strings (via pins 1, 8, and 14).



Theory of operation:

The heart of the Color Organ is the LM324 Quad Op Amp IC, with three of the four amps in the IC configured to function as audio filters. An Op Amp is basically a modular amplifier in an integrated circuit ("IC") package, with most Op Amps designed to provide extremely high amplification ("gain"). In most cases, max gain is typically well above a factor of 10,000 (gain is usually expressed in spec sheets in "decibels" or "dB"; where 3dB = 2x, 10dB = 10x, 20dB = 100x, 40dB = 10,000x, etc.). Most basic Op Amps typically have two inputs: the inverting input ("–") (i.e. any audio signal applied to this input is amplified and *flipped* before it reaches the output), and the non-inverting input ("+") (i.e. any signal applied to this input is amplified but not inverted).

Since Op Amps come pre-designed with the full gain already built in, we must reduce that gain down to the level we need by feeding back some of the signal from the output into the inverting input. Gain is inversely related to the feedback current (I_{FB}). In other words, the smaller the feedback resistor, the more signal current we feed back, and therefore the more we *reduce* the gain.

If instead of using only a resistor in the feedback path, we place some frequency sensitive components (e.g. a capacitor, which "blocks" low frequencies while offering a decreasing impedance to progressively higher frequencies; see appendix), we can make the Op Amp selectively amplify a certain range of frequencies while rejecting others. In other words, we can turn the Op Amp into a frequency selective filter. A simple "low-pass" filter example is shown in Figure 2. This circuit will amplify frequencies up to about 1600 Hz (cycles per second), while frequencies higher than 1600 Hz will be attenuated. We can increase that cutoff frequency by decreasing the "R*C time constant" (i.e. by reducing either the capacitor or the resistor value) in the feedback loop.

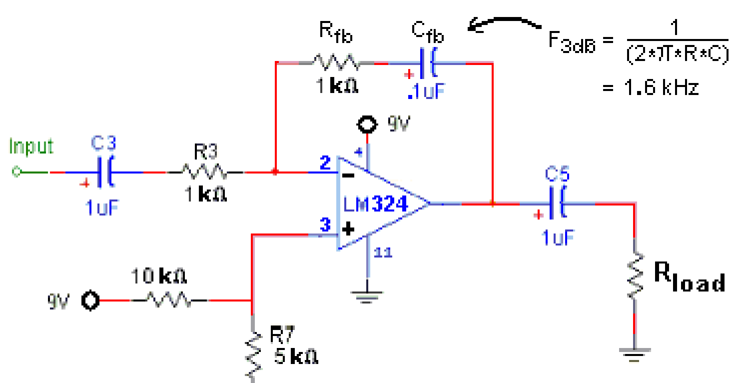
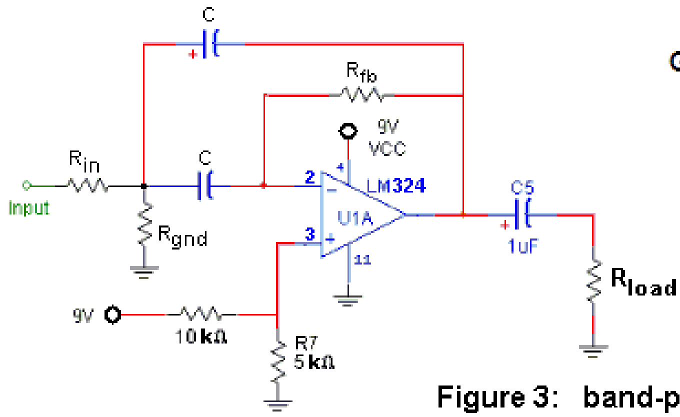


Figure 2: simple low pass filter

In the filters used in the Color Organ circuit, both the input paths and the feedback paths are frequency selective, allowing us to affect frequencies above and below the filter's center frequency, making the filter a "band-pass" filter. By adjusting the R*C component values used in each path, we can change a number of characteristics of these band-pass filters, including their center frequency, their bandwidth, and their gain.

One problem that may affect performance is the level of audio signal injected into the circuit. If it is too loud or any of the amplifiers apply too much gain, the signal will become distorted. These distortions will manifest as additional frequency components (e.g. over-driving a low frequency tone, say from a drum, will generate distortions that will appear as the MF Yellow and HF Red LEDs flashing when no high frequency audio is present in the music). Though this is obviously not a fatal problem, if you encounter it and wonder why, now you know. To mitigate this problem, turn down the input signal and if it still persists, you can adjust the Op Amp gains (increasing Rin will lower the gain, but will also affect the Fcntr).



Given: A_V (gain), F_{cntr}, BW, Q (= F_{cntr}/BW)

$R_{fb} = Q / (2 \text{ Pi } F_{cntr} C)$

$R_{in} = R_{fb} / 2 A_V$

$R_{gnd} = (R_{fb}/2) / (2Q^2 - A_V)$

Figure 3: band-pass filter

<u>Troubleshooting</u>:

Treating the overall project as a collection of <u>*five*</u> distinct sections simplifies the troubleshooting by allowing us to use any symptoms to point to the problem section(s). For example, are all of the sections dead? If so, you probably have a problem in a section they all have in common (e.g. the Power Supply, the Audio Input/pre-amp, etc.). Otherwise, you have screwed up something in all three sections (e.g. flipped diodes around in all LED strings),

One of the most common failure points in any circuit is the supply voltage. In fact, if the circuit worked previously but is now dead, 90% of the time the problem is either with the supply voltage or the ground. First verify the voltage on the chip is good with a DVM. If not, check the output of the Power Supply section, and if there is no voltage at the output of the Power Supply, check the raw input supply voltage, then the ground lines, etc.

If the chip has power but the LED's are still not lighting up, verify that the Audio pre-amp section feeding all three sections is supplying a good signal into each of the filter sections (with an oscilloscope, a DVM, or an earphone). If it is not, check the audo signal feeding into the Audio pre-amp section. If the input audio is good, but nothing is coming out pin 7, check your soldering work and the components around pins 5, 6, and 7, etc.

If there is a strong audio signal on the output of the filter amps (pins 1, 8, and/or 14), verify that you haven't flipped one or more of the LEDs around in the dead LED string(s), etc. If you tested and marked each LED *before* soldering (as suggested previously), placing a bad LED in the string, and/or installing the LED backwards is much less likely (but not impossible).

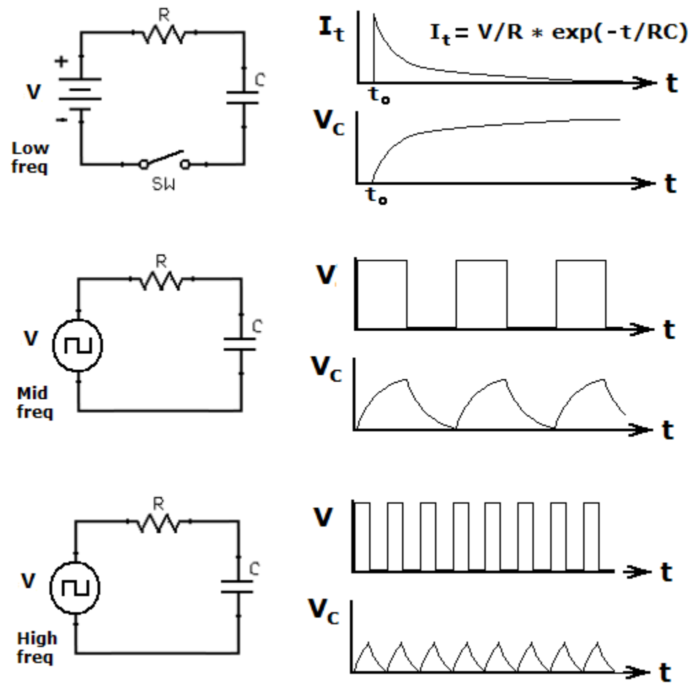
<u>Conclusions</u>:

And that's pretty much it. Hope you enjoy the light show. Check back often, as we expect to be adding additional projects to our website (EpiphanyBySteveLee.com) as time goes by.

Happy trails.

Appendix: additional reference information

Capacitors:



Though capacitors are relatively simple components, their operation still tends to be somewhat mysterious to many hobbyists and technicians. So a word or two on how they work might help explain their key role in our Op Amp filter design.

A capacitor is created anytime two current carrying conductors or "plates" are placed next to each other, separated by some kind of insulator (e.g. plastic, mica, air, Teflon, etc.). When a voltage is applied (e.g. when the switch is closed in the top circuit at the right), current flows from the battery ("V"), through the resistor, and onto the top capacitor plate. As charges accumulate on the top plate, their electrostatic field repels charges of the same polarity on the opposite plate, pushing them out of the bottom of the capacitor (due to the rule: "like charges repel"). For the first instant in time, this flow of charges ("current") is at its maximum (limited only by R, as dictated by Ohm's law: I = V/R). Current however drops off over time "t" at an exponential rate, according to: I(t) = V/R * exp(–t/RC), as it becomes harder and harder for our applied voltage to repel any additional charges from the bottom plate. *At no time do the charges entering the capacitor at the top plate travel through the insulator and out the bottom plate* (unless a high voltage is applied and they arc across, and thus ruin the capacitor).

At that first instant when voltage is applied and current starts to flow, the voltage difference across the capacitor is zero, since the plates start off with the same (balanced) charge. However as charges are pushed off the bottom plate, the difference in charge (and therefore in voltage, V = Q * C) across the two plates increases. In other words, the capacitor offers little impedance to current flow at that first instant, but as the supply of available charges on the bottom plate dwindles, the electrostatic field's ability to force these charges on the bottom plate to exit likewise dwindles, and thus the capacitor's opposition to current flow (i.e. its impedance, "Z") increases over time.

Soon the impedance of the capacitor is much greater than the impedance of the resistor, causing the bulk of the voltage being applied to the circuit to appear across the capacitor (again according to Ohm's law, Vc = I * Zc). When the electrostatic force has depleted all "available" charges off the bottom plate, the capacitor's impedance is roughly equivalent to an open circuit (i.e. extremely high impedance), causing the current in the circuit to drop to zero. At this point, all of the voltage being applied to the circuit is across the capacitor, while the voltage across the resistor drops to zero (since Vr = I * R = 0 * R = 0).

The *rate* at which capacitors charge up (i.e. how quickly they reach full charge) depends on two things: 1) the size of the capacitor "C" itself, and 2) the size of the resistor "R" in series slowing the flow of current into the cap. The above exponential equation for current suggests that a capacitor will charge up to 68% of the applied voltage in one "time constant" (where 1 TC = R*C), 95% of full charge in two time constants, 99% of full charge in three, etc.

Since capacitors present a large impedance when charged and low frequencies give the capacitor plenty of time to charge up, their impedance to low frequencies is very high. Conversely, since high frequency signals alternate polarity very quickly, a capacitor never has a chance to fully charge up when high frequency signals are passed through it (see the bottom circuit in the figure above), giving them very little impedance to high frequencies. Hence the general description used in most introductory discussions on capacitors that they “block low frequencies, and pass high frequencies.”

Applying all of this to our Op Amp filter discussion, we note that when we place a resistor and capacitor in series in the feedback path of any Op Amp (see Figure 2), we create a frequency dependent filter that offers a very small impedance to any high frequency signals feeding back to the inverting input. Feeding back all the high frequencies into the inverting input effectively kills all such high frequencies moving through the Op Amp. At the same time, this filter presents a large impedance in the feedback path for low frequency signals, keeping the low frequencies out of the inverting input and thus allowing the Op Amp to apply its high gain to these low frequencies.

<u>The Five key “rules” for Op Amp design</u>:

There are five general characteristics or “rules” that define the operation of a theoretically “ideal” Op Amp:

1) The ideal Op Amp provides “Infinite” gain.
2) The two input pins (V_+ and V_-) follow each other (i. e. $V_+ = V_-$) via the feedback path.
3) Input current is zero (i.e. the Op Amp input pins draw virtually no current).
4) Output current is “infinite”.
5) The V_+ input pin impedance is infinite, while the output impedance is zero.

As an example, we will use these rules to derive the equation that describes the output for our low-pass filter shown in Figure 2:

Let "s" = j 2 π F. With $I_{IN} = 0$, $V_{FB} = V_{out}$, and thus $I_{IN} = I_{FB}$:

Vin / Rin = Vfb / (Rfb + 1/s C), multiply through by Rin / Vfb.

Vin / Vout = Rin / (Rfb + 1/s C)

Since voltage gain (“Av”) = Vout / Vin, we invert this last line to find Av for our low-pass filter:

$$Av = Vout/Vin = [\,Rfb + 1/(\,j\,2\pi F C)\,] / Rin = [\,Rfb - j\,Xc\,] / Rin$$

where: $Xc = 1/(2\pi F C)$

and: $1/j = -j$