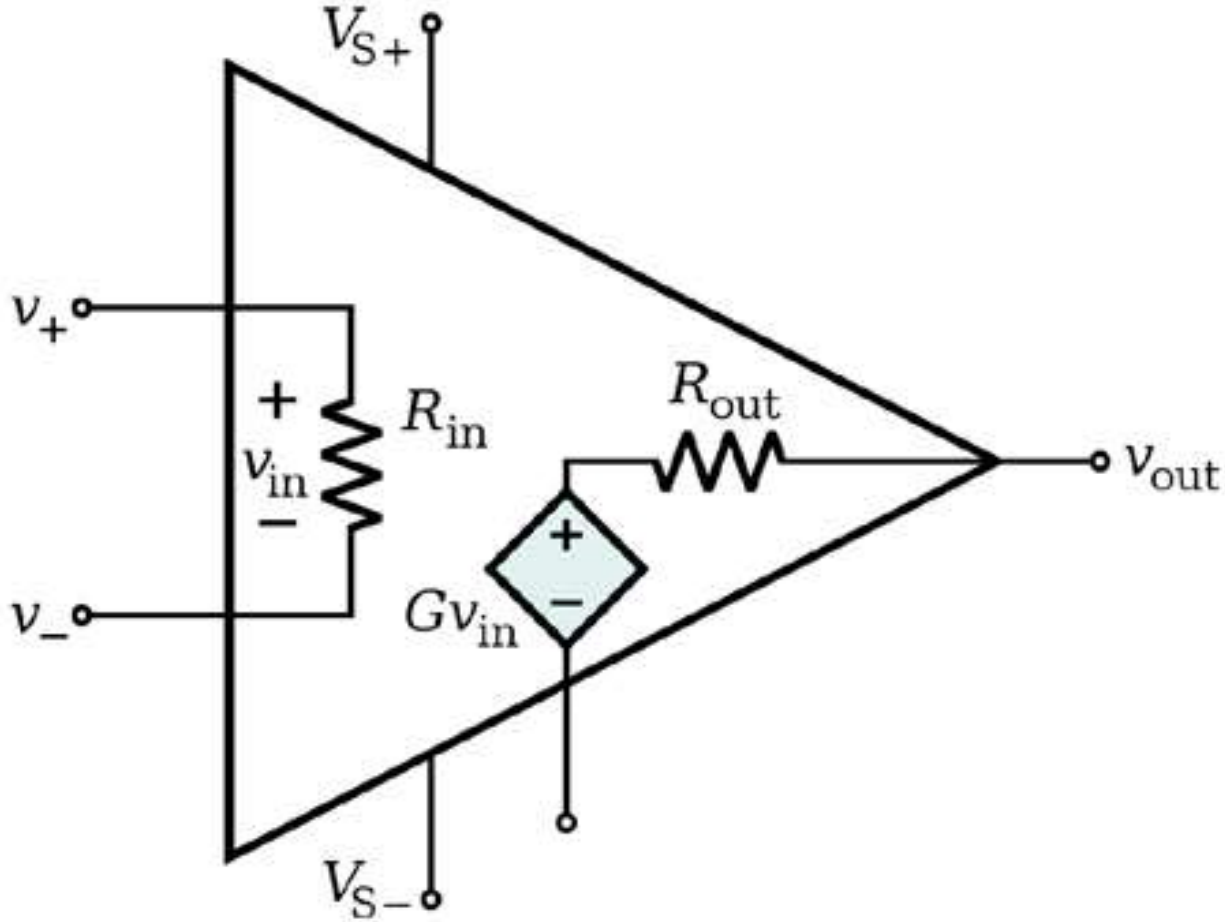
Linear Integrated Circuits, Electronic Amplifier & Electronic Filter Topology



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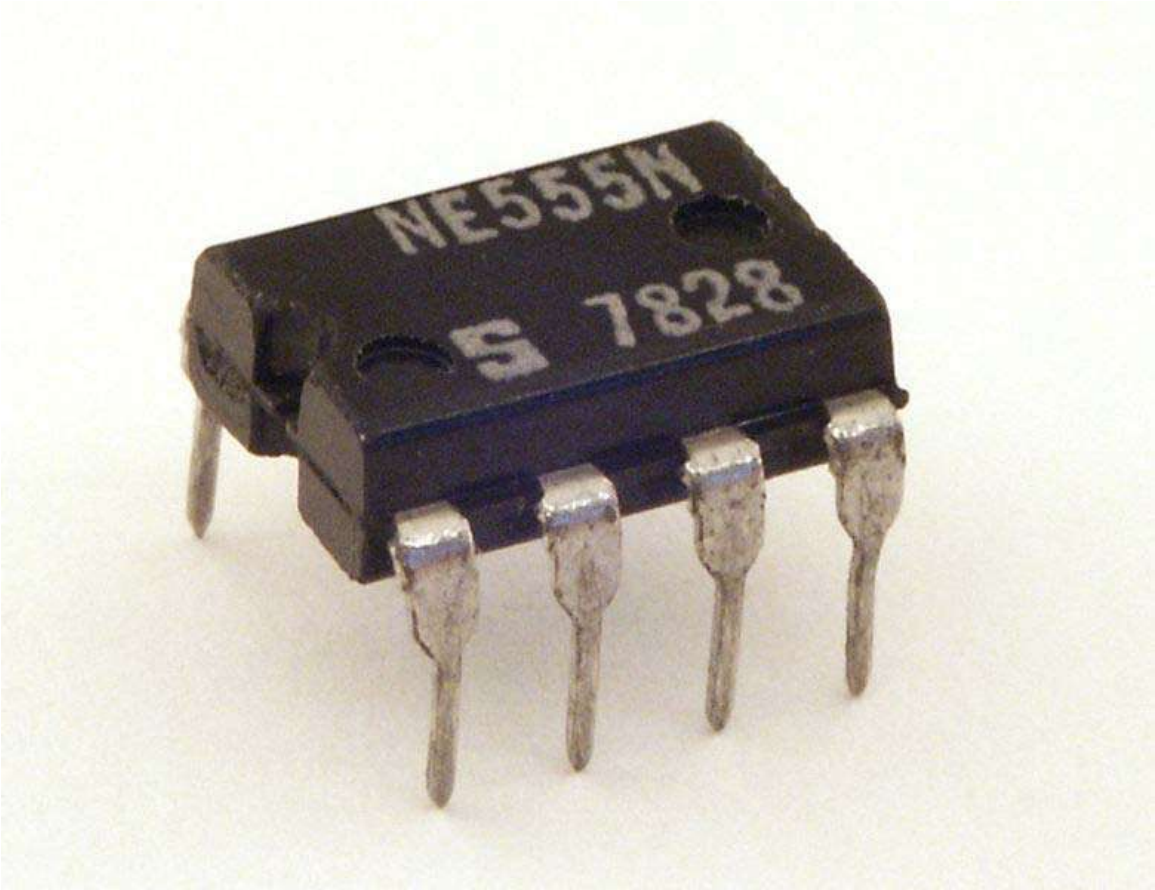
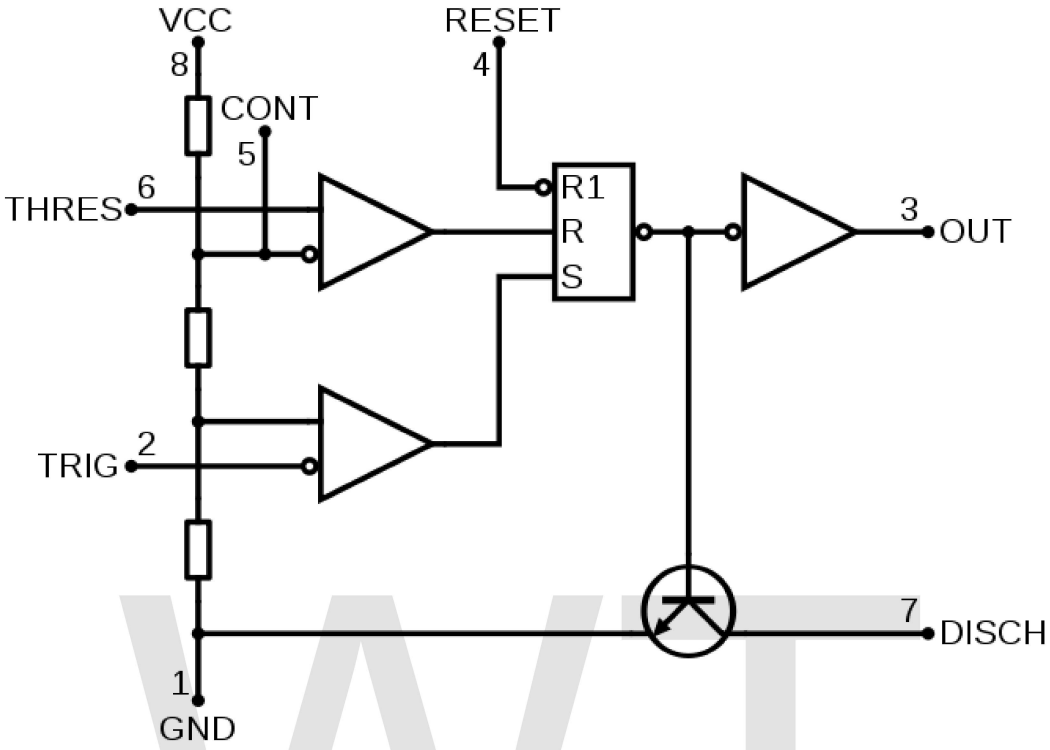
555 Timer IC



NE555 from Signetics in dual-in-line package



Internal block diagram

The **555 Timer IC** is an integrated circuit (chip) used in a variety of timer, pulse generation and oscillator applications. The IC was designed by Hans R. Camenzind in 1970 and brought to market in 1971 by Signetics (later acquired by Philips). The original name was the SE555 (metal can)/**NE555** (plastic DIP) and the part was described as "The IC Time Machine". It has been claimed that the 555 gets its name from the three 5 kΩ resistors used in typical early implementations, but Hans Camenzind has stated that the number was arbitrary. The part is still in wide use, thanks to its ease of use, low price and good stability. As of 2003, it is estimated that 1 billion units are manufactured every year.

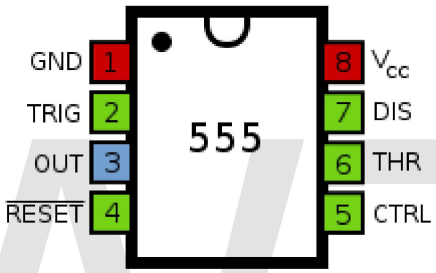
Depending on the manufacturer, the standard 555 package includes over 20 transistors, 2 diodes and 15 resistors on a silicon chip installed in an 8-pin mini dual-in-line package (DIP-8). Variants available include the 556 (a 14-pin DIP combining two 555s on one chip), and the 558 (a 16-pin DIP combining four slightly modified 555s with DIS & THR connected internally, and TR falling edge sensitive instead of level sensitive).

Ultra-low power versions of the 555 are also available, such as the 7555 and TLC555. The 7555 is designed to cause less supply glitching than the classic 555 and the manufacturer claims that it usually does not require a "control" capacitor and in many cases does not require a power supply bypass capacitor.

The 555 has three operating modes:

- Monostable mode: in this mode, the 555 functions as a "one-shot". Applications include timers, missing pulse detection, bouncefree switches, touch switches, frequency divider, capacitance measurement, pulse-width modulation (PWM) etc
- Astable - free running mode: the 555 can operate as an oscillator. Uses include LED and lamp flashers, pulse generation, logic clocks, tone generation, security alarms, pulse position modulation, etc.
- Bistable mode or Schmitt trigger: the 555 can operate as a flip-flop, if the DIS pin is not connected and no capacitor is used. Uses include bouncefree latched switches, etc.

Usage

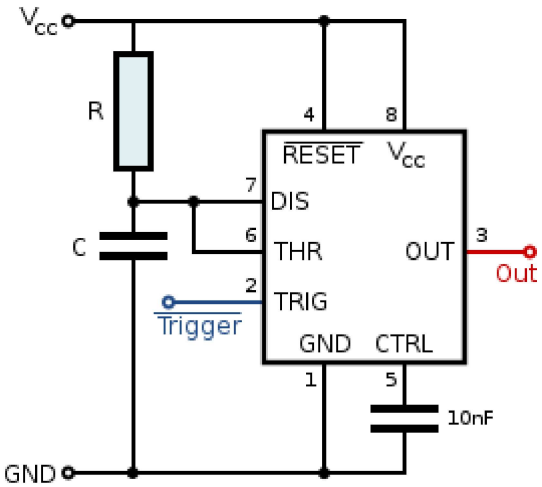


Pinout diagram

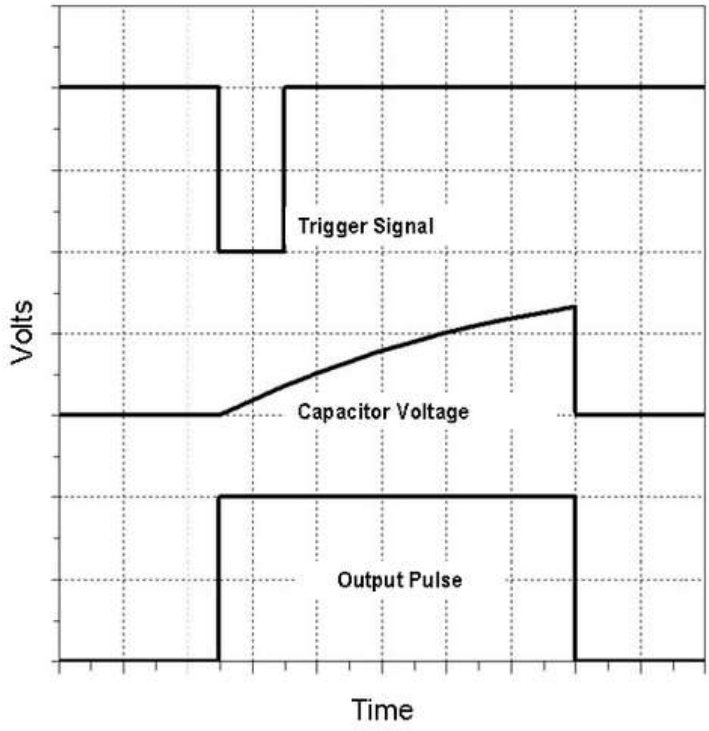
The connection of the pins is as follows:

Pin	Name	Purpose
1	GND	Ground, low level (0 V)
2	TRIG	OUT rises, and interval starts, when this input falls below 1/3 V_{CC}.
3	OUT	This output is driven to $+V_{CC}$ or GND.
4	RESET	A timing interval may be interrupted by driving this input to GND.
5	CTRL	"Control" access to the internal voltage divider (by default, 2/3 V_{CC}).
6	THR	The interval ends when the voltage at THR is greater than at CTRL.
7	DIS	Open collector output; may discharge a capacitor between intervals.
8	V+, V_{CC}	Positive supply voltage is usually between 3 and 15 V.

Monostable mode



Schematic of a 555 in monostable mode



The relationships of the trigger signal, the voltage on C and the pulse width in monostable mode

In the monostable mode, the 555 timer acts as a “one-shot” pulse generator. The pulse begins when the 555 timer receives a signal at the trigger input that falls below a third of the voltage supply. The width of the output pulse is determined by the time constant of an RC network, which consists of a capacitor (C) and a resistor (R). The output pulse ends when the charge on the C equals 2/3 of the supply voltage. The output pulse width can be lengthened or shortened to the need of the specific application by adjusting the values of R and C.

The output pulse width of time *t*, which is the time it takes to charge C to 2/3 of the supply voltage, is given by

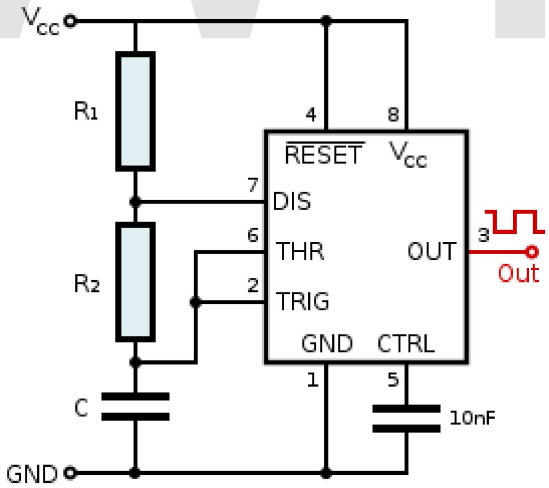
$$t = RC\ln(3) \approx 1.1RC$$

where t is in seconds, R is in ohms and C is in farads.

Bistable Mode

In bistable mode, the 555 timer acts as a basic flip-flop. The trigger and reset inputs (pins 2 and 4 respectively on a 555) are held high via Pull-up resistors while the threshold input (pin 6) is simply grounded. Thus configured, pulling the trigger momentarily to ground acts as a 'set' and transitions the output pin (pin 3) to Vcc (high state). Pulling the reset input to ground acts as a 'reset' and transitions the output pin to ground (low state). No capacitors are required in a bistable configuration. Pins 5 and 7 (control and discharge) are left floating.

Astable mode



Standard 555 Astable Circuit

In astable mode, the 555 timer puts out a continuous stream of rectangular pulses having a specified frequency. Resistor R_1 is connected between V_{CC} and the discharge pin (pin 7) and another resistor (R_2) is connected between the discharge pin (pin 7), and the trigger

(pin 2) and threshold (pin 6) pins that share a common node. Hence the capacitor is charged through R_1 and R_2, and discharged only through R_2, since pin 7 has low impedance to ground during output low intervals of the cycle, therefore discharging the capacitor.

In the astable mode, the frequency of the pulse stream depends on the values of R_1, R_2 and C:

$$f = \frac{1}{\ln(2) \cdot C \cdot (R_1 + 2R_2)}$$

The high time from each pulse is given by

$$\text{high} = \ln(2) \cdot (R_1 + R_2) \cdot C$$

and the low time from each pulse is given by

$$\text{low} = \ln(2) \cdot R_2 \cdot C$$

where R_1 and R_2 are the values of the resistors in ohms and C is the value of the capacitor in farads.

note: power of R_1 must be greater than $\frac{V_{cc}^2}{R_1}$

To achieve a duty cycle of less than 50% a diode can be added in parallel with R_2 towards the capacitor. This bypasses R_2 during the high part of the cycle so that the high interval depends only on R_1 and C.

Specifications

These specifications apply to the NE555. Other 555 timers can have different specifications depending on the grade (military, medical, etc).

Supply voltage (V_{CC})	4.5 to 15 V
Supply current (V_{CC} = +5 V)	3 to 6 mA
Supply current (V_{CC} = +15 V)	10 to 15 mA
Output current (maximum)	200 mA
Maximum Power dissipation	600 mW
Power Consumption (minimum operating)	30 mW@5V, 225 mW@15V
Operating temperature	0 to 70 °C

Derivatives

Many pin-compatible variants, including CMOS versions, have been built by various companies. Bigger packages also exist with two or four timers on the same chip. The 555 is also known under the following type numbers:

Manufacturer	Model	Remark
Avago Technologies	Av-555M	
Custom Silicon Solutions	CSS555/CSS555C	CMOS from 1.2 V, IDD < 5 µA
ECG Philips	ECG955M	
Exar	XR-555	
Fairchild Semiconductor	NE555/KA555	
Harris	HA555	
IK Semicon	ILC555	CMOS from 2 V
Intersil	SE555/NE555	
Intersil	ICM7555	CMOS
Lithic Systems	LC555	
Maxim	ICM7555	CMOS from 2 V
Motorola	MC1455/MC1555	
National Semiconductor	LM1455/LM555/LM555C	
National Semiconductor	LMC555	CMOS from 1.5 V
NTE Sylvania	NTE955M	
Raytheon	RM555/RC555	
RCA	CA555/CA555C	
STMicroelectronics	NE555N/ K3T647	
Texas Instruments	SN52555/SN72555	
Texas Instruments	TLC555	CMOS from 2 V
USSR	К1006ВИ1	
Zetex	ZSCT1555	down to 0.9 V
NXP Semiconductors	ICM7555	CMOS
HFO / East Germany	B555	

Dual timer 556

The dual version is called 556. It features two complete 555s in a 14 pin DIL package.

Quad timer 558

The quad version is called 558 and has 16 pins. To fit four 555s into a 16 pin package the control, voltage, and reset lines are shared by all four modules. Also for each module the discharge and threshold are internally wired together and called *timing*.

Example applications

Joystick interface circuit using quad timer 558

The Apple II microcomputer used a quad timer 558 in monostable (or "one-shot") mode to interface up to four "game paddles" or two joysticks to the host computer.

A similar circuit was used in the IBM personal computer. In the joystick interface circuit of the IBM PC, the capacitor (C) of the RC network was generally a 10 nF capacitor. The resistor (R) of the RC network consisted of the potentiometer inside the joystick along with an external resistor of 2.2 kilohms. The joystick potentiometer acted as a variable resistor. By moving the joystick, the resistance of the joystick increased from a small value up to about 100 kilohms. The joystick operated at 5 V.

Software running in the host computer started the process of determining the joystick position by writing to a special address (ISA bus I/O address 201h). This would result in a trigger signal to the quad timer, which would cause the capacitor (C) of the RC network to begin charging and cause the quad timer to output a pulse. The width of the pulse was determined by how long it took the C to charge up to 2/3 of 5 V (or about 3.33 V), which was in turn determined by the joystick position.

Software running in the host computer measured the pulse width to determine the joystick position. A wide pulse represented the full-right joystick position, for example, while a narrow pulse represented the full-left joystick position.

Atari Punk Console

One of Forrest M. Mims III's many books was dedicated to the 555 timer. In it, he first published the "Stepped Tone Generator" circuit which has been adopted as a popular circuit, known as the Atari Punk Console, by circuit benders for its distinctive low-fi sound similar to classic Atari games.

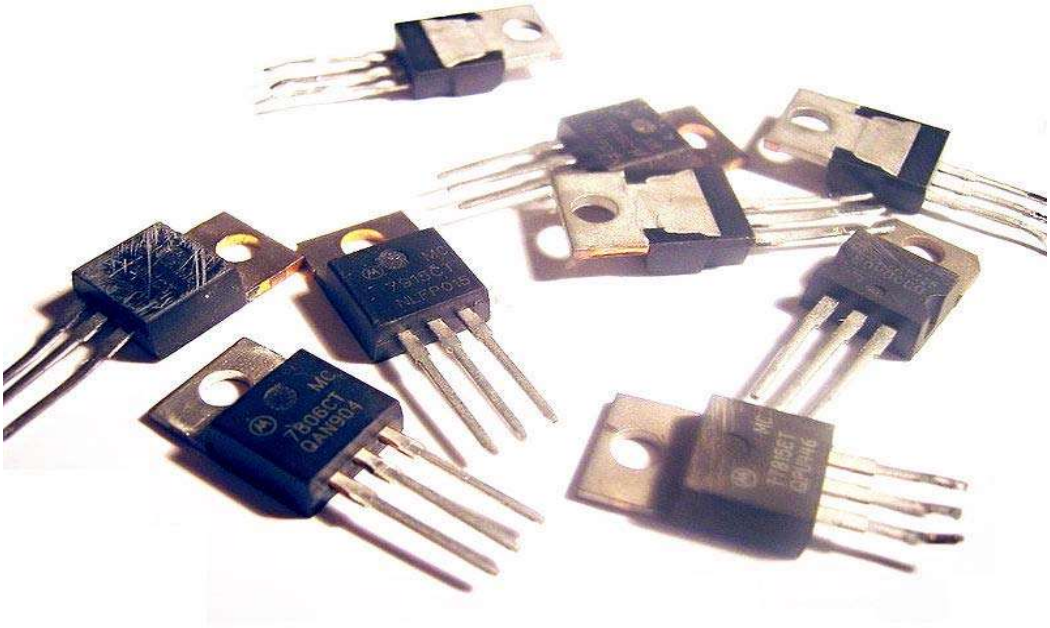
Pulse-width modulation

The 555 can be used to generate a variable PWM signal using a few external components. The chip alone can drive small external loads or an amplifying transistor for larger loads.

Chapter 2

78xx & ZN414

78xx



An assortment of 78XX ICs

The **78xx** (sometimes **LM78xx**) is a family of self-contained fixed linear voltage regulator integrated circuits. The 78xx family is commonly used in electronic circuits requiring a regulated power supply due to their ease-of-use and low cost. For ICs within the family, the *xx* is replaced with two digits, indicating the output voltage (for example, the 7805 has a 5 volt output, while the 7812 produces 12 volts). The 78xx line are positive voltage regulators: they produce a voltage that is positive relative to a common ground. There is a related line of **79xx** devices which are complementary negative voltage regulators. 78xx and 79xx ICs can be used in combination to provide positive and negative supply voltages in the same circuit.

78xx ICs have three terminals and are commonly found in the TO220 form factor, although smaller surface-mount and larger TO3 packages are available. These devices support an input voltage anywhere from a couple of volts over the intended output voltage, up to a maximum of 35 or 40 volts, and typically provide 1 or 1.5 amps of current (though smaller or larger packages may have a lower or higher current rating).

Advantages

- 78xx series ICs do not require additional components to provide a constant, regulated source of power, making them easy to use, as well as economical and efficient uses of space. Other voltage regulators may require additional components to set the output voltage level, or to assist in the regulation process. Some other designs (such as a switching power supply) may need substantial engineering expertise to implement.

- 78xx series ICs have built-in protection against a circuit drawing too much power. They have protection against overheating and short-circuits, making them quite robust in most applications. In some cases, the current-limiting features of the 78xx devices can provide protection not only for the 78xx itself, but also for other parts of the circuit.

Disadvantages

- The input voltage must always be higher than the output voltage by some minimum amount (typically 2 volts). This can make these devices unsuitable for powering some devices from certain types of power sources (for example, powering a circuit that requires 5 volts using 6-volt batteries will not work using a 7805).

- As they are based on a linear regulator design, the input current required is always the same as the output current. As the input voltage must always be higher than the output voltage, this means that the total power (voltage multiplied by current) going into the 78xx will be more than the output power provided. The extra input power is dissipated as heat. This means both that for some applications an adequate heatsink must be provided, and also that a (often substantial) portion of the input power is wasted during the process, rendering them less efficient than some other types of power supplies. When the input voltage is significantly higher than the regulated output voltage (for example, powering a 7805 using a 24 volt power source), this inefficiency can be a significant issue.

- Even in larger packages, 78xx integrated circuits cannot supply as much power as many designs which use discrete components, and are generally inappropriate for applications requiring more than a few amps of current.

Individual Devices in the Series

There are common configurations for 78xx ICs, including 7805 (5 volt), 7806 (6 volt), 7808 (8 volt), 7809 (9 volt), 7810 (10 volt), 7812 (12 volt), 7815 (15 volt), 7818 (18 volt), and 7824 (24 volt) versions. The 7805 is common, as its regulated 5 volt supply provides a convenient power source for most TTL components.

Less common are lower-power versions such as the LM78Mxx series (500mA) and LM78Lxx series (100mA) from National Semiconductor. Some devices provide slightly different voltages than usual, such as the LM78L62 (6.2 volts) and LM78L82 (8.2 volts).

Unrelated Devices

The LM78S40 from National Semiconductor is not part of the 78xx family, and does not use the same design. It is a component in switching regulator designs, and is not a linear regulator like other 78xx devices. The 7803SR from Datel is a full switching power supply module (designed as a drop-in replacement for 78xx chips), and not a linear regulator like the 78xx ICs.

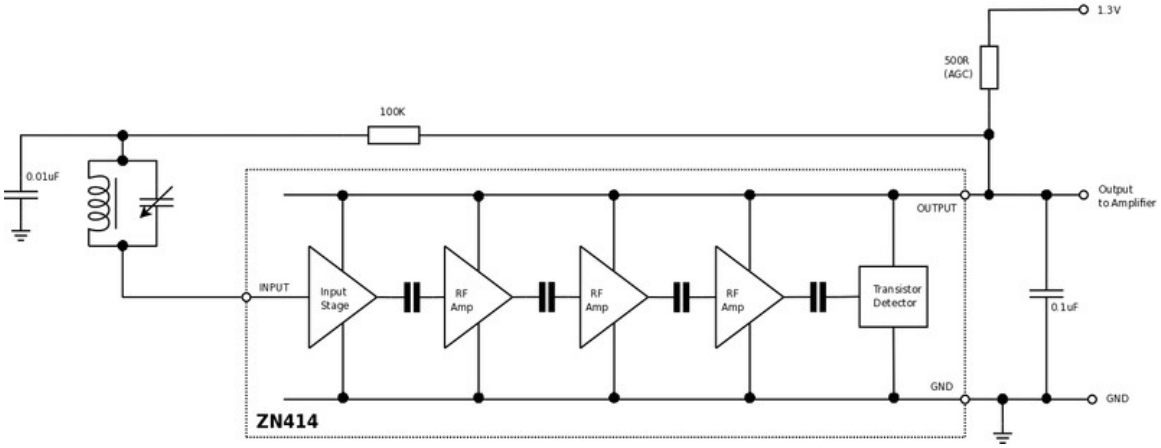
ZN414

ZN414 in TO92 package

The ZN414 was a low cost, single-chip AM radio integrated circuit. Launched in 1972, the part was designed and supplied by Ferranti, but was also available from GEC-Plessey. The ZN414 was popular amongst hobbyists as a fully working AM radio could be made with just a few external components, a crystal earpiece and a 1.5 V cell.

The original ZN414 chip from Ferranti was supplied in a 3-pin, metal TO-18 'transistor' package whereas the GEC part and later Ferranti ones (ZN414Z) used the plastic TO-92 encapsulation. Later variants, the ZN415 and ZN416, came in 8-pin DIL packages and included a built-in amplifier that could drive headphones and small speakers directly.

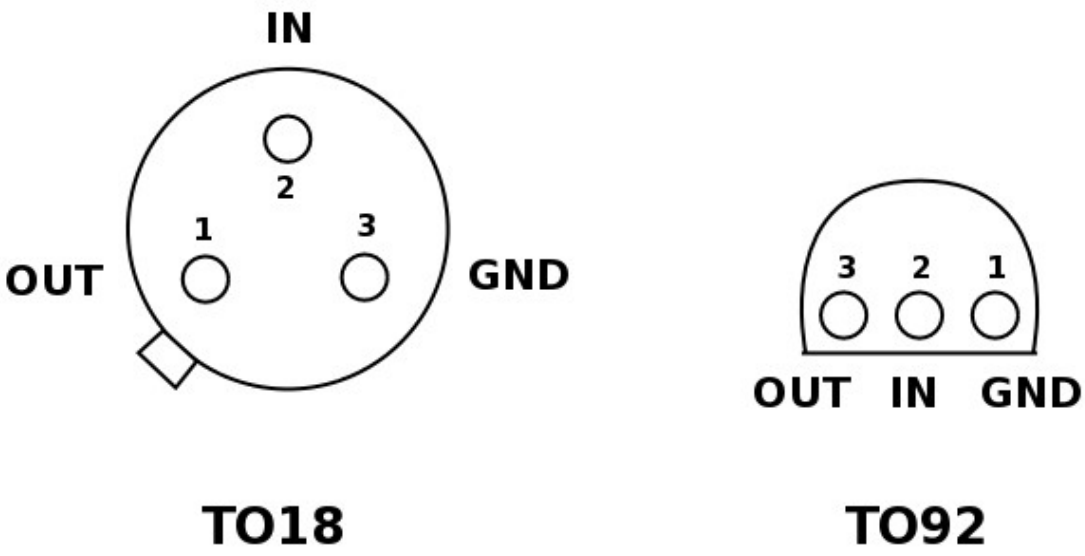
The radio circuit inside the ZN414 was based on a design known as Tuned Radio Frequency (TRF). The TRF design is much simpler than the popular, but more complex, superheterodyne radio circuit often used in modern AM receivers. It was principally the use of the TRF circuit that allowed almost a whole radio to be fitted into one small, three-pin package.



ZN414 in basic functional circuit

The manufacturing process for the ZN414 chip used a relatively new (for the time) technique known as Collector Diffusion Isolation (CDI). CDI was invented by engineers at Bell Telephone Laboratories and subsequently developed into a commercial process by Ferranti in the UK.

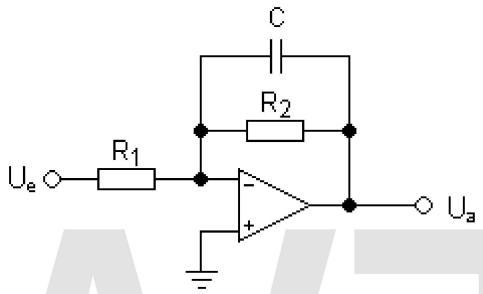
The original ZN41x family have not been manufactured for some time, but modern equivalents to the original 3-pin ZN414 are available, with part codes of MK484, TA7642 and (mainly in India, the Far East & Australasia) YS414 and LMF501T. Note that on the YS414 part, pins 1 (output) and 3 (ground/earth) are transposed.



ZN414 pinouts

Chapter 3

Electronic Filter Topology



An elementary filter topology introduces a capacitor into the feedback path of an op-amp to achieve an unbalanced active implementation of a low-pass transfer function

Electronic filter topology defines electronic filter circuits without taking note of the values of the components used but only the manner in which those components are connected.

Filter design characterises filter circuits primarily by their transfer function rather than their topology. Transfer functions may be linear or nonlinear. Common types of linear filter transfer function are; high-pass, low-pass, bandpass, band-reject or notch and all-pass. Once the transfer function for a filter is chosen the particular topology to implement such a prototype filter can be selected so that, for example, one might choose to design a Butterworth filter using the Sallen–Key topology.

Filter topologies may be divided into passive and active types. Passive topologies do not contain a generator of energy, either in reality or, due to non-linearity, in their equivalent circuit, but only capacitors, inductors and also, in some topologies, resistors: particularly, variable resistors are often included in order to control the depth of filtering. Active topologies include components that require power. Further, topologies may be implemented either in unbalanced form or else in balanced form when employed in balanced circuits. Implementations such as electronic mixers and stereo sound may require arrays of identical circuits.

Passive topologies

Passive filters have been long in development and use. Most are built from simple two-port networks called "sections". There is no formal definition of a section except that it must have at least one series component and one shunt component. Sections are invariably connected in a "cascade" or "daisy-chain" topology, consisting of either repeats of the same section or of completely different sections. Impedance would combine two sections consisting only of series components or shunt components into a single section.

Some passive filters, consisting of only one or two filter sections, are given special names including the L-section, T-section and Π-section, which are unbalanced filters, and the C-section, H-section and box-section, which are balanced. All are built upon a very simple "ladder" topology (see below). The chart at the bottom of the page shows these various topologies in terms of general constant k filters.

Filters designed using network synthesis usually repeat the simplest form of L-section topology though component values may change in each section. Image designed filters, on the other hand, keep the same basic component values from section to section though the topology may vary and tend to make use of more complex sections.

L-sections are never symmetrical but two L-sections back-to-back form a symmetrical topology and many other sections are symmetrical in form.

Ladder topologies

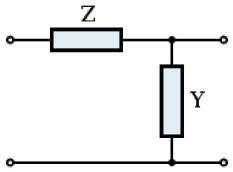
Ladder topology, often called **Cauer topology** after Wilhelm Cauer (inventor of the Elliptical filter), was in fact first used by George Campbell (inventor of the Constant k filter). Campbell published in 1922 but had clearly been using the topology for some time before this. Cauer first picked up on ladders (published 1926) inspired by the work of Foster (1924). There are two forms of basic ladder topologies; unbalanced and balanced. Cauer topology is usually thought of as an unbalanced ladder topology.

A ladder network consists of cascaded asymmetrical L-sections (unbalanced) or C-sections (balanced). In low pass form the topology would consist of series inductors and shunt capacitors. Other bandforms would have an equally simple topology transformed from the lowpass topology. The transformed network will have shunt admittances that are dual networks of the series impedances if they were duals in the starting network - which is the case with series inductors and shunt capacitors.

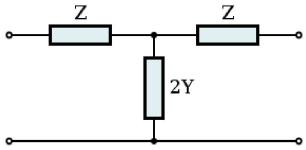
Image filter sections

Unbalanced

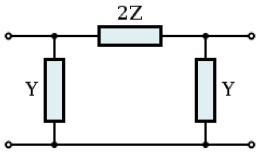
L Half section



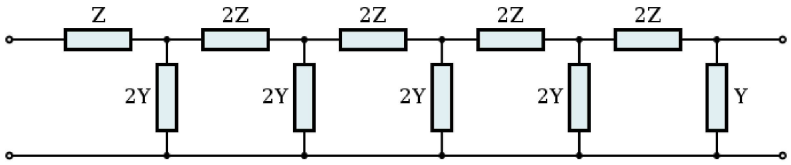
T Section



Π Section

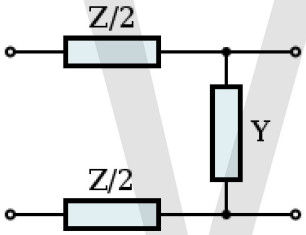


Ladder network

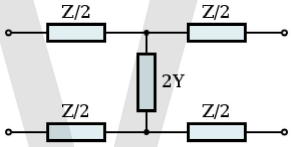


Balanced

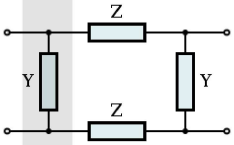
C Half-section



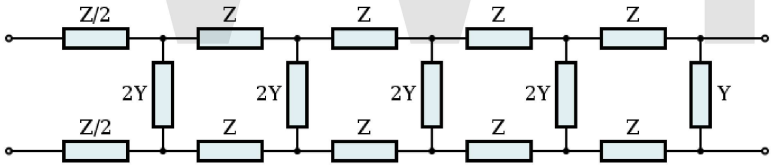
H Section



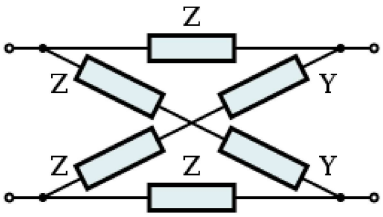
Box Section



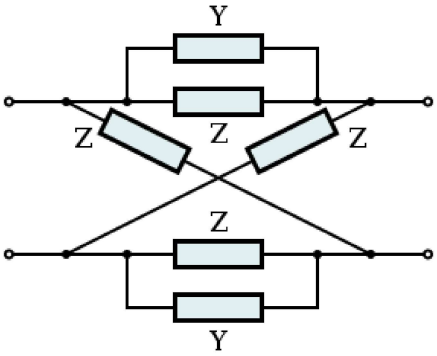
Ladder network



X Section (mid-T-Derived)



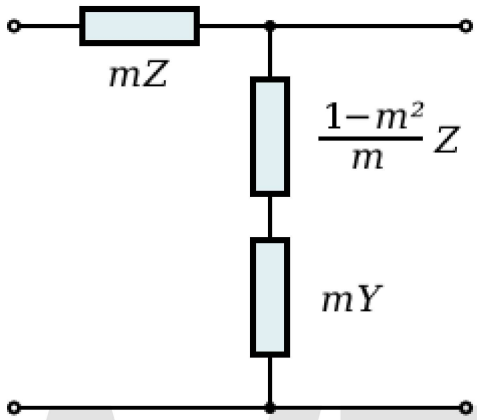
X Section (mid-Π-Derived)



N.B. Textbooks and design drawings usually show the unbalanced implementations, but in telecoms it is often required to convert the design to the balanced

implementation when used with balanced lines.

Modified ladder topologies



series m-derived topology

Image filter design commonly uses modifications of the basic ladder topology. These topologies, invented by Otto Zobel, have the same passbands as the ladder on which they are based but their transfer functions are modified to improve some parameter such as impedance matching, stopband rejection or passband-to-stopband transition steepness. Usually the design applies some transform to a simple ladder topology: the resulting topology is ladder-like but no longer obeys the rule that shunt admittances are the dual network of series impedances: it invariably becomes more complex with higher component count. Such topologies include;

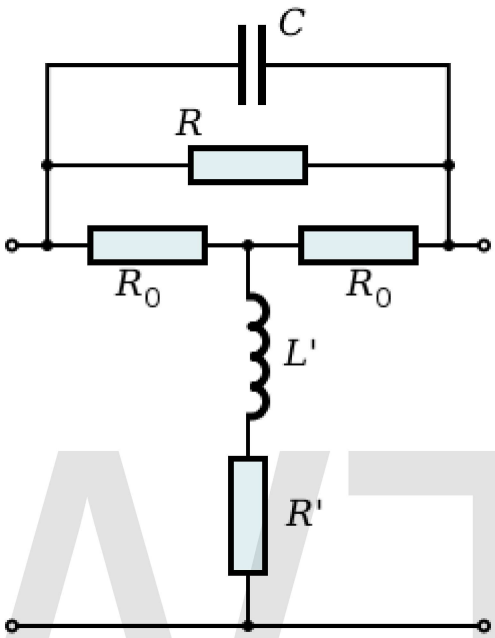
- m-derived filter
- mm'-type filter
- General m_n-type filter

The m-type (m-derived) filter is by far the most commonly used modified image ladder topology. There are two m-type topologies for each of the basic ladder topologies; the series-derived and shunt-derived topologies. These have identical transfer functions to each other but different image impedances. Where a filter is being designed with more than one passband, the m-type topology will result in a filter where each passband has an analogous frequency-domain response. It is possible to generalise the m-type topology for filters with more than one passband using parameters m_1, m_2, m_3 etc., which are not equal to each other resulting in general m_n-type filters which have bandforms that can differ in different parts of the frequency spectrum.

The mm'-type topology can be thought of as a double m-type design. Like the m-type it has the same bandform but offers further improved transfer characteristics. It is, however,

a rarely used design due to increased component count and complexity as well as its normally requiring basic ladder and m-type sections in the same filter for impedance matching reasons. It is normally only found in a composite filter.

Bridged-T topologies



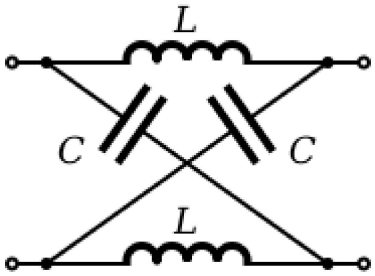
Typical bridged-T Zobel network equaliser used to correct high end roll-off

Zobel constant resistance filters use a topology that is somewhat different from other filter types, distinguished by having a constant input resistance at all frequencies and in that they use resistive components in the design of their sections. The higher component and section count of these designs usually limits their use to equalisation applications. Topologies usually associated with constant resistance filters are the bridged-T and its variants, all described in the Zobel network article;

- Bridged-T topology
- Balanced bridged-T topology
- Open-circuit L-section topology
- Short-circuit L-section topology
- Balanced open-circuit C-section topology
- Balanced short-circuit C-section topology

The bridged-T topology is also used in sections intended to produce a signal delay but in this case no resistive components are used in the design.

Lattice topology



Lattice topology X-section phase correction filter

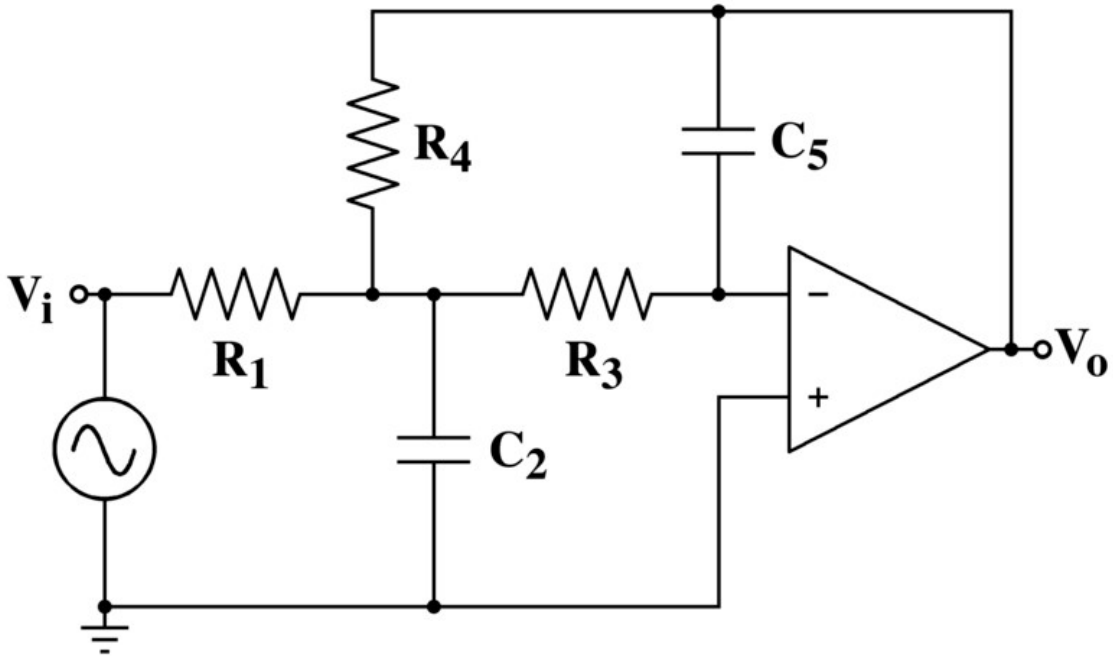
Both the T-section (from ladder topology) and the bridge-T (from Zobel topology) can be transformed into a lattice topology filter section but in both cases this results in high component count and complexity. The most common application of lattice filters (X-sections) is in all-pass filters used for phase equalisation.

Although T and bridged-T sections can always be transformed into X-sections the reverse is not always possible because of the possibility of negative values of inductance and capacitance arising in the transform.

Lattice topology is identical to the more familiar **bridge topology**, the difference being merely the drawn representation on the page rather than any real difference in topology, cicuitry or function.

Active topologies

Multiple feedback topology



Multiple feedback topology circuit

Multiple feedback topology is an electronic filter topology which is used to implement an electronic filter by adding two poles to the transfer function. A diagram of the circuit topology for a second order low pass filter is shown in the figure on the right.

The transfer function of the multiple feedback topology circuit, like all second-order linear filters, is:

$$H(s) = \frac{V_o}{V_i} = -\frac{1}{As^2 + Bs + C} = \frac{K\omega_0^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}.$$

In an MF filter,

$$A = (R_1 R_3 C_2 C_5)$$
$$B = R_3 C_5 + R_1 C_5 + R_1 R_3 C_5 / R_4$$
$$C = R_1 / R_4$$
$$Q = \frac{\sqrt{R_3 R_4 C_2 C_5}}{(R_4 + R_3 + |K| R_3) C_5}$$ is the Q factor.
$$K = -R_4 / R_1$$ is the DC voltage gain

$\omega_0 = 2\pi f_0 = 1/\sqrt{R_3 R_4 C_2 C_5}$ is the corner frequency

Biquad filter

A **biquad filter** is a type of linear filter that implements a transfer function that is the ratio of two quadratic functions. The name *biquad* is short for *biquadratic*.

Biquad filters are typically active and implemented with a **single-amplifier biquad (SAB)** or **two-integrator-loop** topology.

- The SAB topology uses feedback to generate complex poles and possibly complex zeros. In particular, the feedback moves the real poles of an RC circuit in order to generate the proper filter characteristics.
- The two-integrator-loop topology is derived from rearranging a biquadratic transfer function. The rearrangement will equate one signal with the sum of another signal, its integral, and the integral's integral. In other words, the rearrangement reveals a state variable filter structure. By using different states as outputs, any kind of second-order filter can be implemented.

The SAB topology is sensitive to component choice and can be more difficult to adjust. Hence, usually the term **biquad** refers to the two-integrator-loop state variable filter topology.

Tow-Thomas Biquad Example

For example, the basic configuration in Figure 1 can be used as either a low-pass or bandpass filter depending on where the output signal is taken from.

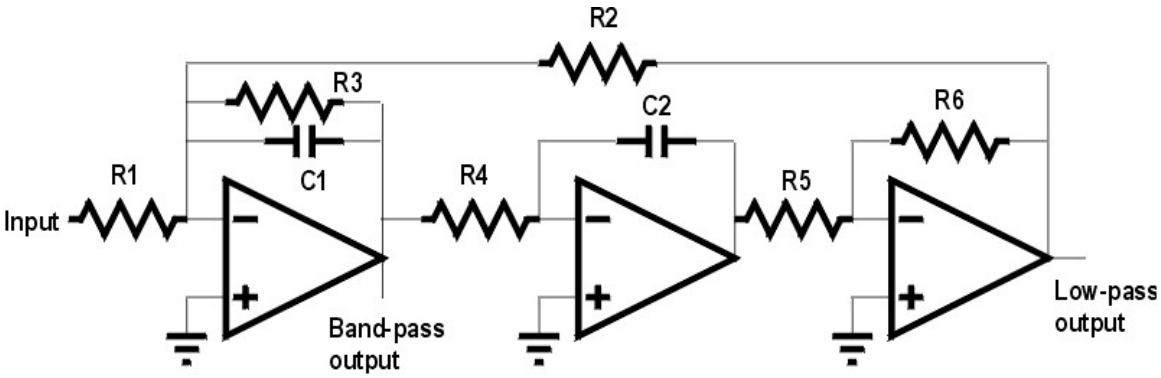


Figure 1: The common Tow-Thomas biquad filter topology.

The second-order low-pass transfer function is given by

$$H(s) = \frac{G_{lpf}\omega_0^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}$$

where low-pass gain $G_{lpf} = R_2 / R_1$. The second-order bandpass transfer function is given by

$$H(s) = \frac{G_{bpf} \frac{\omega_0}{Q} s}{s^2 + \frac{\omega_0}{Q} s + \omega_0^2}.$$

with bandpass gain $G_{bpf} = -R_4 / R_2$. In both cases, the

- Natural frequency is $\omega_0 = 1/\sqrt{R_2 R_4 C_1 C_2}$.
- Quality factor is $Q = \sqrt{\frac{R_3^2 C_1}{R_2 R_4 C_2}}$.

The bandwidth is approximated by $B = \omega_0 / Q$, and Q is sometimes expressed as a damping constant $\zeta = 1 / 2Q$. If a noninverting low-pass filter is required, the output can be taken at the output of the second operational amplifier. If a noninverting bandpass filter is required, the order of the second integrator and the inverter can be switched, and the output taken at the output of the inverter's operational amplifier.

Chapter 4

RC Circuit

A **resistor–capacitor circuit (RC circuit)**, or **RC filter** or **RC network**, is an electric circuit composed of resistors and capacitors driven by a voltage or current source. A first order RC circuit is composed of one resistor and one capacitor and is the simplest type of RC circuit.

RC circuits can be used to filter a signal by blocking certain frequencies and passing others. The four most common RC filters are the high-pass filter, low-pass filter, band-pass filter, and band-stop filter.

Introduction

There are three basic, linear passive lumped analog circuit components: the resistor (R), capacitor (C) and inductor (L). These may be combined in: the RC circuit, the RL circuit, the LC circuit and the RLC circuit with the abbreviations indicating which components are used. These circuits, between them, exhibit a large number of important types of behaviour that are fundamental to much of analog electronics. In particular, they are able to act as passive filters.

Natural response

The simplest RC circuit is a capacitor and a resistor in series. When a circuit consists of only a charged capacitor and a resistor, the capacitor will discharge its stored energy through the resistor. The voltage across the capacitor, which is time dependent, can be found by using Kirchhoff's current law, where the current through the capacitor must equal the current through the resistor. This results in the linear differential equation

$$C\frac{dV}{dt} + \frac{V}{R} = 0.$$

Solving this equation for V yields the formula for exponential decay:

$$V(t) = V_0 e^{-\frac{t}{RC}},$$

where V_0 is the capacitor voltage at time $t = 0$.

$$\frac{V_0}{e}$$

The time required for the voltage to fall to $\frac{V_0}{e}$ is called the RC time constant and is given by

$$\tau = RC\ .$$

Complex impedance

The complex impedance, Z_C (in ohms) of a capacitor with capacitance C (in farads) is

$$Z_C = \frac{1}{sC}$$

The complex frequency s is, in general, a complex number,

$$s = \sigma + j\omega$$

where

- j represents the imaginary unit:

  $j^2 = -1$

- σ is the exponential decay constant (in radians per second), and
- ω is the sinusoidal angular frequency (also in radians per second).

Sinusoidal steady state

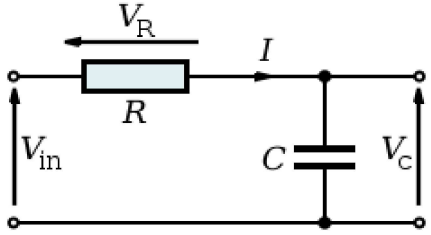
Sinusoidal steady state is a special case in which the input voltage consists of a pure sinusoid (with no exponential decay). As a result,

$$\sigma = 0$$

and the evaluation of s becomes

$$s = j\omega$$

Series circuit



Series RC circuit

By viewing the circuit as a voltage divider, the voltage across the capacitor is:

$$V_C(s) = \frac{1/Cs}{R + 1/Cs} V_{in}(s) = \frac{1}{1 + RCs} V_{in}(s)$$

and the voltage across the resistor is:

$$V_R(s) = \frac{R}{R + 1/Cs} V_{in}(s) = \frac{RCs}{1 + RCs} V_{in}(s).$$

Transfer functions

The transfer function for the capacitor is

$$H_C(s) = \frac{V_C(s)}{V_{in}(s)} = \frac{1}{1 + RCs}.$$

Similarly, the transfer function for the resistor is

$$H_R(s) = \frac{V_R(s)}{V_{in}(s)} = \frac{RCs}{1 + RCs}.$$

Poles and zeros

Both transfer functions have a single pole located at

$$s = -\frac{1}{RC}.$$

In addition, the transfer function for the resistor has a zero located at the origin.

Gain and phase angle

The magnitude of the gains across the two components are:

$$G_C = |H_C(j\omega)| = \left|\frac{V_C(j\omega)}{V_{in}(j\omega)}\right| = \frac{1}{\sqrt{1 + (\omega RC)^2}}$$

and

$$G_R = |H_R(j\omega)| = \left|\frac{V_R(j\omega)}{V_{in}(j\omega)}\right| = \frac{\omega RC}{\sqrt{1+(\omega RC)^2}},$$

and the phase angles are:

$$\phi_C = \angle H_C(j\omega) = \tan^{-1}(-\omega RC)$$

and

$$\phi_R = \angle H_R(j\omega) = \tan^{-1}\left(\frac{1}{\omega RC}\right).$$

These expressions together may be substituted into the usual expression for the phasor representing the output:

$$\begin{aligned} V_C &= G_C V_{in} e^{j\phi_C} \\ V_R &= G_R V_{in} e^{j\phi_R}. \end{aligned}$$

Current

The current in the circuit is the same everywhere since the circuit is in series:

$$I(s) = \frac{V_{in}(s)}{R + \frac{1}{Cs}} = \frac{Cs}{1+RCs} V_{in}(s)$$

Impulse response

The impulse response for each voltage is the inverse Laplace transform of the corresponding transfer function. It represents the response of the circuit to an input voltage consisting of an impulse or Dirac delta function.

The impulse response for the capacitor voltage is

$$h_C(t) = \frac{1}{RC} e^{-t/RC} u(t) = \frac{1}{\tau} e^{-t/\tau} u(t)$$

where $u(t)$ is the Heaviside step function and

$$\tau = RC$$

is the time constant.

Similarly, the impulse response for the resistor voltage is

$$h_R(t) = \delta(t) - \frac{1}{RC} e^{-t/RC} u(t) = \delta(t) - \frac{1}{\tau} e^{-t/\tau} u(t)$$

where $\delta(t)$ is the Dirac delta function

Frequency-domain considerations

These are frequency domain expressions. Analysis of them will show which frequencies the circuits (or filters) pass and reject. This analysis rests on a consideration of what happens to these gains as the frequency becomes very large and very small.

As $\omega \to \infty$:

$$\begin{aligned} G_C &\to 0 \\ G_R &\to 1. \end{aligned}$$

As $\omega \to 0$:

$$\begin{aligned} G_C &\to 1 \\ G_R &\to 0. \end{aligned}$$

This shows that, if the output is taken across the capacitor, high frequencies are attenuated (rejected) and low frequencies are passed. Thus, the circuit behaves as a *low-pass filter*. If, though, the output is taken across the resistor, high frequencies are passed and low frequencies are rejected. In this configuration, the circuit behaves as a *high-pass filter*.

The range of frequencies that the filter passes is called its bandwidth. The point at which the filter attenuates the signal to half its unfiltered power is termed its cutoff frequency. This requires that the gain of the circuit be reduced to

$$G_C = G_R = \frac{1}{\sqrt{2}}.$$

Solving the above equation yields

$$\omega_c = \frac{1}{RC} \text{ rad/s}$$

or

$$f_c = \frac{1}{2\pi RC} \text{ Hz}$$

which is the frequency that the filter will attenuate to half its original power.

Clearly, the phases also depend on frequency, although this effect is less interesting generally than the gain variations.

As $\omega \to 0$:

$$\begin{aligned}\phi_C &\to 0 \\ \phi_R &\to 90^\circ = \pi/2^c.\end{aligned}$$

As $\omega \to \infty$:

$$\begin{aligned}\phi_C &\to -90^\circ = -\pi/2^c \\ \phi_R &\to 0\end{aligned}$$

So at DC (0 Hz), the capacitor voltage is in phase with the signal voltage while the resistor voltage leads it by 90°. As frequency increases, the capacitor voltage comes to have a 90° lag relative to the signal and the resistor voltage comes to be in-phase with the signal.

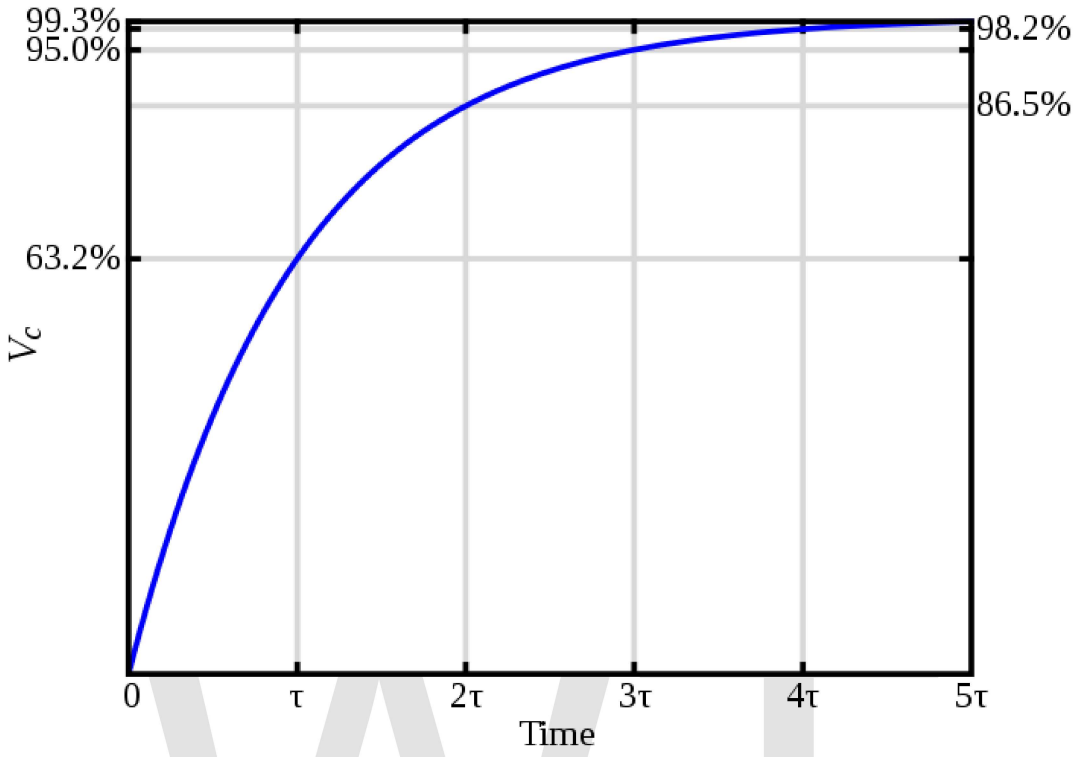
Time-domain considerations

The most straightforward way to derive the time domain behaviour is to use the Laplace transforms of the expressions for V_C and V_R given above. This effectively transforms $j\omega \to s$. Assuming a step input (i.e. $V_{in} = 0$ before $t = 0$ and then $V_{in} = V$ afterwards):

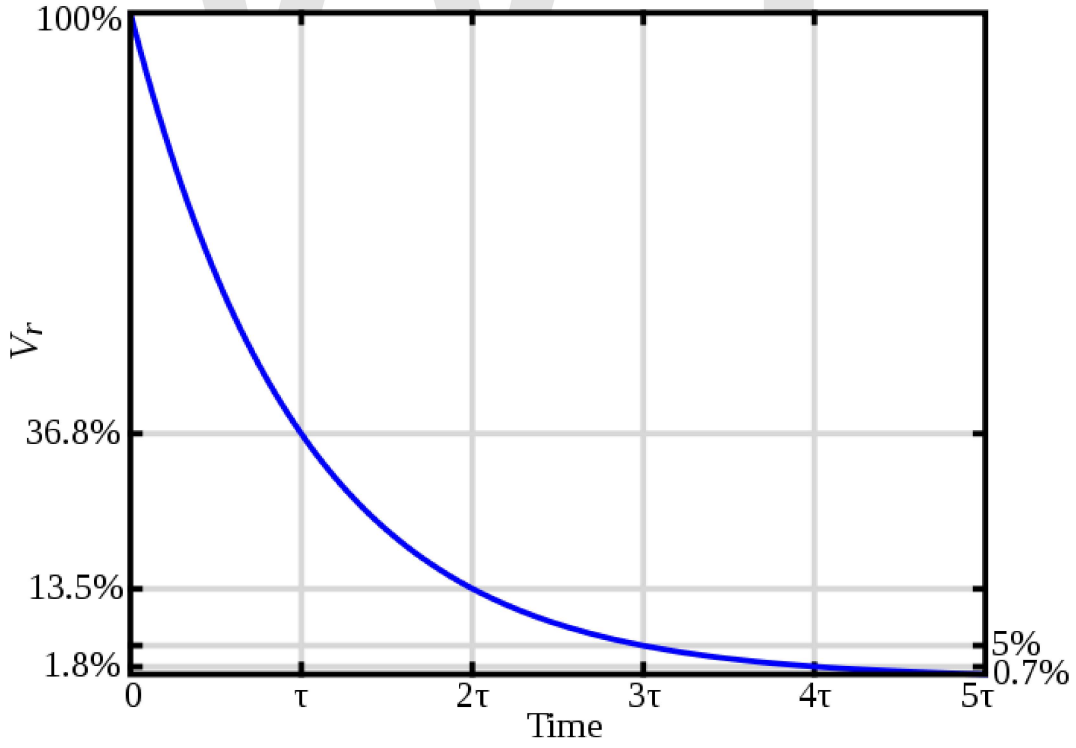
$$\begin{aligned}V_{in}(s) &= V\frac{1}{s} \\ V_C(s) &= V\frac{1}{1+sRC}\frac{1}{s}\end{aligned}$$

and

$$V_R(s) = V\frac{sRC}{1+sRC}\frac{1}{s}.$$



Capacitor voltage step-response



Resistor voltage step-response

Partial fractions expansions and the inverse Laplace transform yield:

$$V_C(t) = V\left(1 - e^{-t/RC}\right)$$
$$V_R(t) = Ve^{-t/RC}.$$

These equations are for calculating the voltage across the capacitor and resistor respectively while the capacitor is charging; for discharging, the equations are vice-versa. These equations can be rewritten in terms of charge and current using the relationships C=Q/V and V=IR.

Thus, the voltage across the capacitor tends towards V as time passes, while the voltage across the resistor tends towards 0, as shown in the figures. This is in keeping with the intuitive point that the capacitor will be charging from the supply voltage as time passes, and will eventually be fully charged and form an open circuit.

These equations show that a series RC circuit has a time constant, usually denoted $\tau = RC$ being the time it takes the voltage across the component to either rise (across C) or fall (across R) to within 1 / e of its final value. That is, τ is the time it takes V_C to reach $V(1 - 1 / e)$ and V_R to reach $V(1 / e)$.

The rate of change is a *fractional* $\left(1 - \frac{1}{e}\right)$ per τ. Thus, in going from $t = N\tau$ to $t = (N + 1)\tau$, the voltage will have moved about 63.2 % of the way from its level at $t = N\tau$ toward its final value. So C will be charged to about 63.2 % after τ, and essentially fully charged (99.3 %) after about 5τ. When the voltage source is replaced with a short-circuit, with C fully charged, the voltage across C drops exponentially with t from V towards 0. C will be discharged to about 36.8 % after τ, and essentially fully discharged (0.7 %) after about 5τ. Note that the current, I, in the circuit behaves as the voltage across R does, via Ohm's Law.

These results may also be derived by solving the differential equations describing the circuit:

$$\frac{V_{in} - V_C}{R} = C\frac{dV_C}{dt}$$

and

$$V_R = V_{in} - V_C.$$

The first equation is solved by using an integrating factor and the second follows easily; the solutions are exactly the same as those obtained via Laplace transforms.

Integrator

Consider the output across the capacitor at *high* frequency i.e.

$$\omega \gg \frac{1}{RC}.$$

This means that the capacitor has insufficient time to charge up and so its voltage is very small. Thus the input voltage approximately equals the voltage across the resistor. To see this, consider the expression for *I* given above:

$$I = \frac{V_{in}}{R + 1/j\omega C}$$

but note that the frequency condition described means that

$$\omega C \gg \frac{1}{R}$$

so

$$I \approx \frac{V_{in}}{R}$$ which is just Ohm's Law.

Now,

$$V_C = \frac{1}{C}\int_0^t I dt$$

so

$$V_C \approx \frac{1}{RC}\int_0^t V_{in} dt,$$

which is an integrator *across the capacitor*.

Differentiator

Consider the output across the resistor at *low* frequency i.e.,

$$\omega \ll \frac{1}{RC}.$$

This means that the capacitor has time to charge up until its voltage is almost equal to the source's voltage. Considering the expression for *I* again, when

$$R \ll \frac{1}{\omega C},$$

so

$$I \approx \frac{V_{in}}{1/j\omega C}$$
$$V_{in} \approx \frac{I}{j\omega C} \approx V_C$$

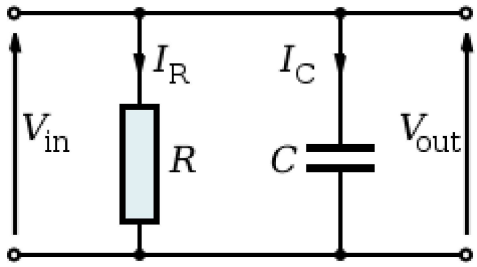
Now,

$$V_R = IR = C\frac{dV_C}{dt}R$$
$$V_R \approx RC\frac{dV_{in}}{dt}$$

which is a differentiator *across the resistor*.

More accurate integration and differentiation can be achieved by placing resistors and capacitors as appropriate on the input and feedback loop of operational amplifiers.

Parallel circuit



Parallel RC circuit

The parallel RC circuit is generally of less interest than the series circuit. This is largely because the output voltage V_{out} is equal to the input voltage V_{in} — as a result, this circuit does not act as a filter on the input signal unless fed by a current source.

With complex impedances:

$$I_R = \frac{V_{in}}{R}$$

and

$$I_C = j\omega C V_{in}.$$

This shows that the capacitor current is 90° out of phase with the resistor (and source) current. Alternatively, the governing differential equations may be used:

$$I_R = \frac{V_{in}}{R}$$

and

$$I_C = C\frac{dV_{in}}{dt}.$$

When fed by a current source, the transfer function of a parallel RC circuit is

$$\frac{V_{out}}{I_{in}} = \frac{R}{1 + sRC}.$$

Chapter 5

RL Circuit

A **resistor-inductor circuit (RL circuit)**, or **RL filter** or **RL network**, is one of the simplest analogue infinite impulse response electronic filters. It consists of a resistor and an inductor, either in series or in parallel, driven by a voltage source.

Introduction

The fundamental passive linear circuit elements are the resistor (R), capacitor (C) and inductor (L). These circuit elements can be combined to form an electrical circuit in four distinct ways: the RC circuit, the RL circuit, the LC circuit and the RLC circuit with the abbreviations indicating which components are used. These circuits exhibit important types of behaviour that are fundamental to analogue electronics. In particular, they are able to act as passive filters.

In practice, however, capacitors (and RC circuits) are usually preferred to inductors since they can be more easily manufactured and are generally physically smaller, particularly for higher values of components.

Complex Impedance

The complex impedance Z_L (in ohms) of an inductor with inductance L (in henries) is

$$Z_L = Ls$$

The complex frequency s is a complex number,

$$s = \sigma + j\omega$$

where

- j represents the imaginary unit:

  $j^2 = -1$

- σ is the exponential decay constant (in radians per second), and

- ω is the angular frequency (in radians per second).

Eigenfunctions

The complex-valued **eigenfunctions** of ANY linear time-invariant (LTI) system are of the following forms:

$V(t) = \mathbf{A}e^{st} = \mathbf{A}e^{(\sigma+j\omega)t}$, or letting $\mathbf{A} = Ae^{j\phi}$ and rewriting;
$= Ae^{j\phi}e^{(\sigma+j\omega)t}$, and collecting terms is $= Ae^{\sigma t}e^{j(\omega t+\phi)}$

From Euler's formula, the **real-part** of these eigenfunctions are exponentially-decaying sinusoids:

$$v(t) = \mathrm{Re}\{V(t)\} = Ae^{\sigma t}\cos(\omega t+\phi)$$

Sinusoidal Steady State

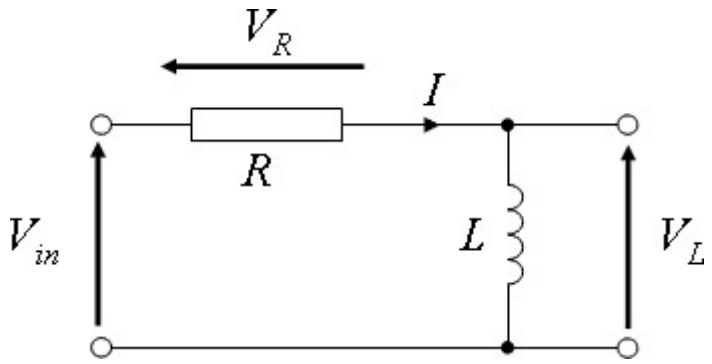
Sinusoidal steady state is a special case in which the input voltage consists of a pure sinusoid (with no exponential decay). As a result,

$$\sigma = 0$$

and the evaluation of *s* becomes

$$s = j\omega$$

Series circuit



Series RL circuit

By viewing the circuit as a voltage divider, we see that the voltage across the inductor is:

$$V_L(s) = \frac{Ls}{R+Ls}V_{in}(s)$$

and the voltage across the resistor is:

$$V_R(s) = \frac{R}{R + Ls} V_{in}(s).$$

Current

The current in the circuit is the same everywhere since the circuit is series:

$$I(s) = \frac{V_{in}(s)}{R + Ls}.$$

Transfer functions

The transfer function for the inductor is

$$H_L(s) = \frac{V_L(s)}{V_{in}(s)} = \frac{Ls}{R + Ls} = G_L e^{j\phi_L}$$

Similarly, the transfer function for the resistor is

$$H_R(s) = \frac{V_R(s)}{V_{in}(s)} = \frac{R}{R + Ls} = G_R e^{j\phi_R}$$

Poles and zeros

Both transfer functions have a single pole located at

$$s = -\frac{R}{L}$$

In addition, the transfer function for the inductor has a zero located at the origin.

Gain and phase angle

The gains across the two components are found by taking the magnitudes of the above expressions:

$$G_L = |H_L(s)| = \left| \frac{V_L(s)}{V_{in}(s)} \right| = \frac{\omega L}{\sqrt{R^2 + (\omega L)^2}}$$

and

$$G_R = |H_R(s)| = \left|\frac{V_R(s)}{V_{in}(s)}\right| = \frac{R}{\sqrt{R^2 + (\omega L)^2}},$$

and the phase angles are:

$$\phi_L = \angle H_L(s) = \tan^{-1}\left(\frac{R}{\omega L}\right)$$

and

$$\phi_R = \angle H_R(s) = \tan^{-1}\left(-\frac{\omega L}{R}\right).$$

Phasor notation

These expressions together may be substituted into the usual expression for the phasor representing the output:

$$V_L = G_L V_{in} e^{j\phi_L}$$
$$V_R = G_R V_{in} e^{j\phi_R}.$$

Impulse Response

The impulse response for each voltage is the inverse Laplace transform of the corresponding transfer function. It represents the response of the circuit to an input voltage consisting of an impulse or Dirac delta function.

The impulse response for the inductor voltage is

$$h_L(t) = \delta(t) - \frac{R}{L}e^{-tR/L}u(t) = \delta(t) - \frac{1}{\tau}e^{-t/\tau}u(t)$$

where $u(t)$ is the Heaviside step function and

$$\tau = \frac{L}{R}$$

is the time constant.

Similarly, the impulse response for the resistor voltage is

$$h_R(t) = \frac{R}{L} e^{-tR/L} u(t) = \frac{1}{\tau} e^{-t/\tau} u(t)$$

Zero input response (ZIR)

The **Zero input response**, also called the **natural response**, of an RL circuit describes the behavior of the circuit after it has reached constant voltages and currents and is disconnected from any power source. It is called the zero-input response because it requires no input.

The ZIR of an RL circuit is:

$$i(t) = i(0)e^{-(R/L)t} = i(0)e^{-t/\tau}.$$

Frequency domain considerations

These are frequency domain expressions. Analysis of them will show which frequencies the circuits (or filters) pass and reject. This analysis rests on a consideration of what happens to these gains as the frequency becomes very large and very small.

As $\omega \to \infty$:

$$\begin{aligned} G_L &\to 1 \\ G_R &\to 0. \end{aligned}$$

As $\omega \to 0$:

$$\begin{aligned} G_L &\to 0 \\ G_R &\to 1. \end{aligned}$$

This shows that, if the output is taken across the inductor, high frequencies are passed and low frequencies are attenuated (rejected). Thus, the circuit behaves as a *high-pass filter*. If, though, the output is taken across the resistor, high frequencies are rejected and low frequencies are passed. In this configuration, the circuit behaves as a *low-pass filter*. Compare this with the behaviour of the resistor output in an RC circuit, where the reverse is the case.

The range of frequencies that the filter passes is called its bandwidth. The point at which the filter attenuates the signal to half its unfiltered power is termed its cutoff frequency. This requires that the gain of the circuit be reduced to

$$G_L = G_R = \frac{1}{\sqrt{2}}.$$

Solving the above equation yields

$$\omega_c = \frac{R}{L} \text{ rad/s}$$

or

$$f_c = \frac{R}{2\pi L} \text{ Hz}$$

which is the frequency that the filter will attenuate to half its original power.

Clearly, the phases also depend on frequency, although this effect is less interesting generally than the gain variations.

As $\omega \to 0$:

$$\begin{aligned} \phi_L &\to 90^\circ = \pi/2^c \\ \phi_R &\to 0 \end{aligned}.$$

As $\omega \to \infty$:

$$\begin{aligned} \phi_L &\to 0 \\ \phi_R &\to -90^\circ = -\pi/2^c \end{aligned}$$

So at DC (0 Hz), the resistor voltage is in phase with the signal voltage while the inductor voltage leads it by 90°. As frequency increases, the resistor voltage comes to have a 90° lag relative to the signal and the inductor voltage comes to be in-phase with the signal.

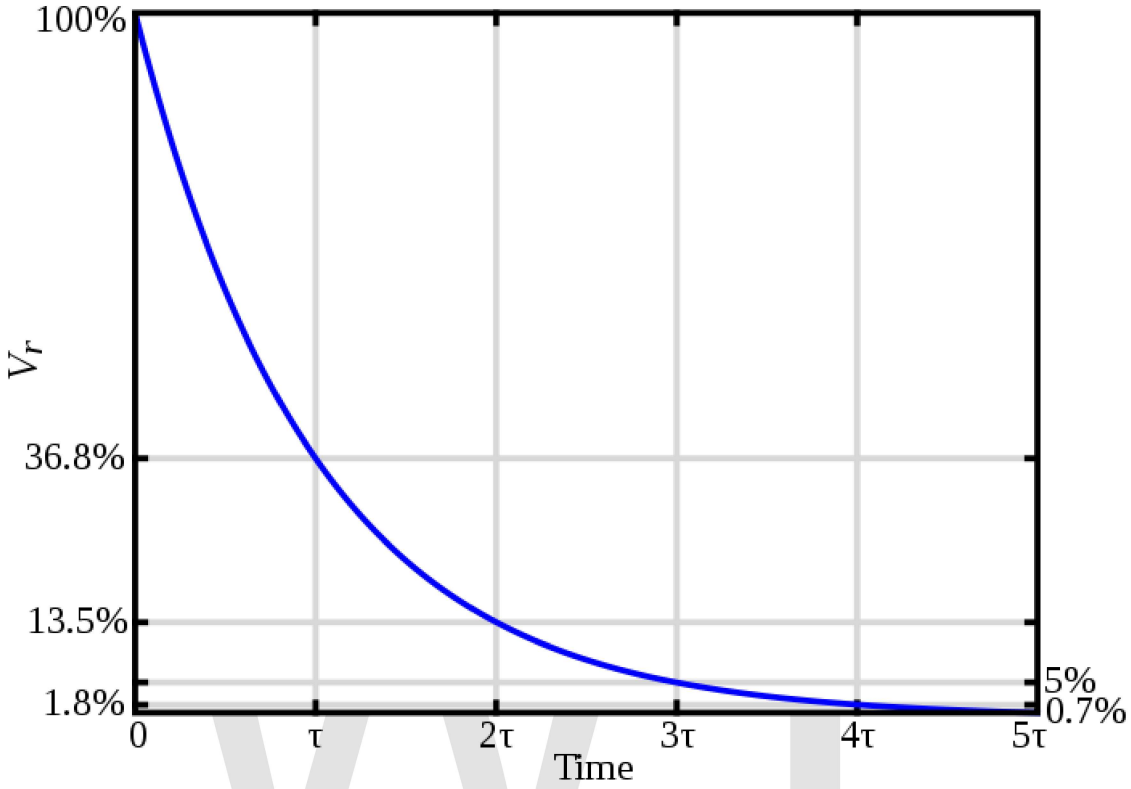
Time domain considerations

The most straightforward way to derive the time domain behaviour is to use the Laplace transforms of the expressions for V_L and V_R given above. This effectively transforms $j\omega \to s$. Assuming a step input (i.e. $V_{in} = 0$ before $t = 0$ and then $V_{in} = V$ afterwards):

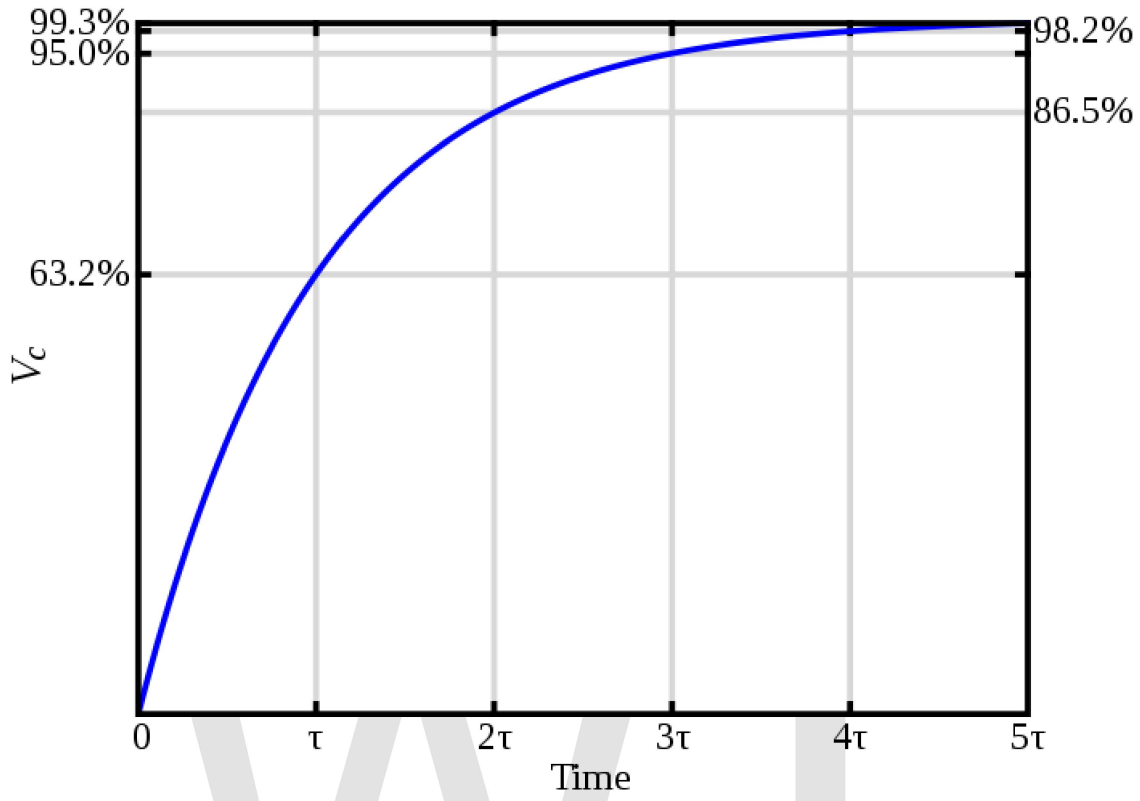
$$\begin{aligned} V_{in}(s) &= V\frac{1}{s} \\ V_L(s) &= V\frac{sL}{R+sL}\frac{1}{s} \end{aligned}$$

and

$$V_R(s) = V\frac{R}{R+sL}\frac{1}{s}.$$



Inductor voltage step-response



Resistor voltage step-response

Partial fractions expansions and the inverse Laplace transform yield:

$$V_L(t) = Ve^{-tR/L}$$
$$V_R(t) = V\left(1 - e^{-tR/L}\right).$$

Thus, the voltage across the inductor tends towards 0 as time passes, while the voltage across the resistor tends towards V, as shown in the figures. This is in keeping with the intuitive point that the inductor will only have a voltage across as long as the current in the circuit is changing — as the circuit reaches its steady-state, there is no further current change and ultimately no inductor voltage.

These equations show that a series RL circuit has a time constant, usually denoted $\tau = L / R$ being the time it takes the voltage across the component to either fall (across L) or rise (across R) to within $1 / e$ of its final value. That is, τ is the time it takes V_L to reach $V(1 / e)$ and V_R to reach $V(1 - 1 / e)$.

The rate of change is a *fractional* $\left(1 - \frac{1}{e}\right)$ per τ. Thus, in going from $t = N\tau$ to $t = (N + 1)\tau$, the votage will have moved about 63% of the way from its level at $t = N\tau$ toward its final value. So the voltage across L will have dropped to about 37% after τ, and

essentially to zero (0.7%) after about 5τ. Kirchhoff's voltage law implies that the voltage across the resistor will *rise* at the same rate. When the voltage source is then replaced with a short-circuit, the voltage across R drops exponentially with t from V towards 0. R will be discharged to about 37% after τ, and essentially fully discharged (0.7%) after about 5τ. Note that the current, I, in the circuit behaves as the voltage across R does, via Ohm's Law.

The delay in the rise/fall time of the circuit is in this case caused by the back-EMF from the inductor which, as the current flowing through it tries to change, prevents the current (and hence the voltage across the resistor) from rising or falling much faster than the time-constant of the circuit. Since all wires have some self-inductance and resistance, all circuits have a time constant. As a result, when the power supply is switched on, the current does not instantaneously reach its steady-state value, V / R. The rise instead takes several time-constants to complete. If this were not the case, and the current were to reach steady-state immediately, extremely strong inductive electric fields would be generated by the sharp change in the magnetic field — this would lead to breakdown of the air in the circuit and electric arcing, probably damaging components (and users).

These results may also be derived by solving the differential equation describing the circuit:

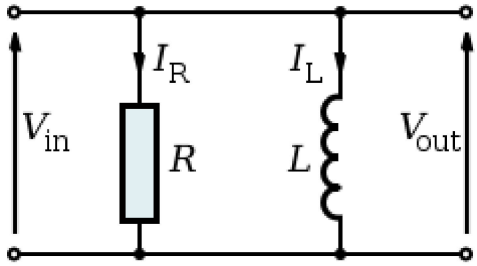
$$V_{in} = IR + L\frac{dI}{dt},$$

and

$$V_R = V_{in} - V_L.$$

The first equation is solved by using an integrating factor and yields the current which must be differentiated to give V_L; the second equation is straightforward. The solutions are exactly the same as those obtained via Laplace transforms.

Parallel circuit



Parallel RL circuit

The parallel RL circuit is generally of less interest than the series circuit unless fed by a current source. This is largely because the output voltage V_{out} is equal to the input voltage V_{in} — as a result, this circuit does not act as a filter for a voltage input signal.

With complex impedances:

$$I_R = \frac{V_{in}}{R}$$

and

$$I_L = \frac{V_{in}}{j\omega L} = -\frac{jV_{in}}{\omega L}.$$

This shows that the inductor lags the resistor (and source) current by 90°.

The parallel circuit is seen on the output of many amplifier circuits, and is used to isolate the amplifier from capacitive loading effects at high frequencies. Because of the phase shift introduced by capacitance, some amplifiers become unstable at very high frequencies, and tend to oscillate. This affects sound quality and component life (especially the transistors), and is to be avoided.

Chapter 6

Sallen–Key Topology

The **Sallen–Key topology** is an electronic filter topology used to implement second-order active filters that is particularly valued for its simplicity. It is a degenerate form of a **voltage-controlled voltage-source (VCVS) filter topology**. A VCVS filter uses a super-unity-gain voltage amplifier with practically infinite input impedance and zero output impedance to implement a 2-pole (12 dB/octave) low-pass, high-pass, or bandpass response. The super-unity-gain amplifier allows for very high Q factor and passband gain without the use of inductors. A Sallen–Key filter is a variation on a VCVS filter that uses a unity-gain amplifier (i.e., a pure buffer amplifier with 0 dB gain). It was introduced by R.P. Sallen and E. L. Key of MIT Lincoln Laboratory in 1955.

Because of its high input impedance and easily selectable gain, an operational amplifier in a conventional non-inverting configuration is often used in VCVS implementations. Implementations of Sallen–Key filters often use an operational amplifier configured as a voltage follower; however, emitter or source followers are other common choices for the buffer amplifier.

VCVS filters are relatively resilient to component tolerance, but obtaining high Q factor may require extreme component value spread or high amplifier gain. Higher-order filters can be obtained by cascading two or more stages.

Generic Sallen–Key topology

The generic unity-gain Sallen–Key filter topology implemented with a unity-gain operational amplifier is shown in Figure 1. The following analysis is based on the assumption that the operational amplifier is ideal.

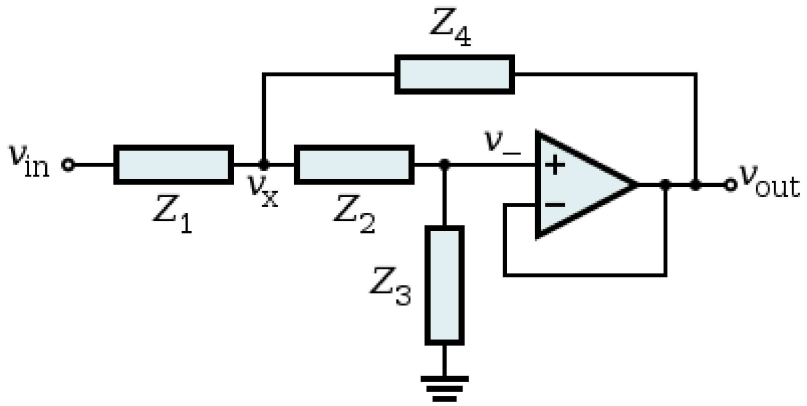


Figure 1: The generic Sallen–Key filter topology.

Because the operational amplifier (OA) is in a negative-feedback configuration, its v_+ and v_- inputs must match (i.e., $v_+ = v_-$). However, the inverting input v_- is connected directly to the output v_{out}, and so

$$v_+ = v_- = v_{\text{out}}. \tag{1}$$

By Kirchhoff's current law (KCL) applied at the v_x node,

$$\frac{v_{\text{in}} - v_x}{Z_1} = \frac{v_x - v_{\text{out}}}{Z_4} + \frac{v_x - v_-}{Z_2}. \tag{2}$$

By combining Equations (1) and (2),

$$\frac{v_{\text{in}} - v_x}{Z_1} = \frac{v_x - v_{\text{out}}}{Z_4} + \frac{v_x - v_{\text{out}}}{Z_2}$$

Applying KCL again at the OA's non-inverting input v_+ (= v_- = v_{out}) gives

$$\frac{v_x - v_{\text{out}}}{Z_2} = \frac{v_{\text{out}}}{Z_3},$$

which means that

$$v_x = v_{\text{out}} \left(\frac{Z_2}{Z_3} + 1 \right).$$

$$(3)$$

Combining Equations (2) and (3) gives

$$\frac{v_{\text{in}} - v_{\text{out}}\left(\frac{Z_2}{Z_3}+1\right)}{Z_1} = \frac{v_{\text{out}}\left(\frac{Z_2}{Z_3}+1\right) - v_{\text{out}}}{Z_4} + \frac{v_{\text{out}}\left(\frac{Z_2}{Z_3}+1\right) - v_{\text{out}}}{Z_2}. \quad (4)$$

Rearranging Equation (4) gives the transfer function

$$\frac{v_{\text{out}}}{v_{\text{in}}} = \frac{Z_3 Z_4}{Z_1 Z_2 + Z_4(Z_1 + Z_2) + Z_3 Z_4}, \quad (5)$$

which typically describes a second-order LTI system.

Interpretation

If the Z_4component were connected to ground, the filter would be a voltage divider composed of the $Z_1$and Z_4components cascaded with another voltage divider composed of the $Z_2$and Z_3components. The buffer bootstraps the "bottom" of the Z_4component to the output of the filter, which will improve upon the simple two divider case. This interpretation is the reason why Sallen–Key filters are often drawn with the operational amplifier's non-inverting input below the inverting input, thus emphasizing the similarity between the output and ground.

Example applications

By choosing different passive components (e.g., resistors and capacitors) for Z_1, Z_2, Z_3, and Z_4, the filter can be made with low-pass, bandpass, and high-pass characteristics. In the examples below, recall that a resistor with resistance Rhas impedance $Z_R$of

$$Z_R = R\,,$$

and a capacitor with capacitance Chas impedance $Z_C$of

$$Z_C = \frac{1}{sC},$$

where $s = j\omega = \left(\sqrt{-1}\right) 2\pi f$ and f is a frequency of a pure sine wave input. That is, a capacitor's impedance is frequency dependent and a resistor's impedance is not.

Example: Low-pass filter

An example of a unity-gain low-pass configuration is shown in Figure 2.

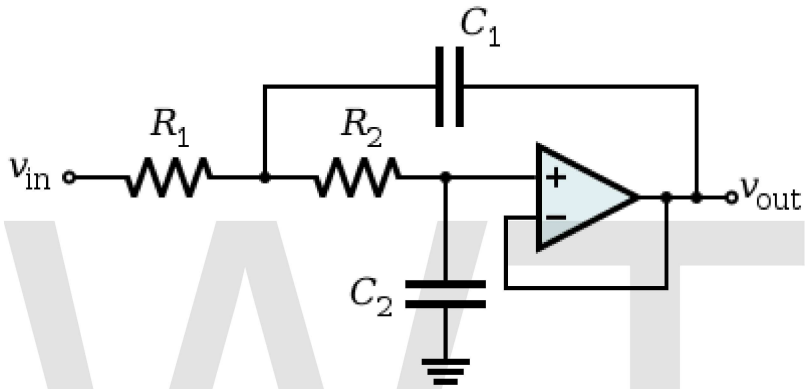


Figure 2: A unity-gain low-pass filter implemented with a Sallen–Key topology.

An operational amplifier is used as the buffer here, although an emitter follower is also effective. This circuit is equivalent to the generic case above with

$$Z_1 = R_1, \quad Z_2 = R_2, \quad Z_3 = \frac{1}{sC_2}, \quad \text{and} \quad Z_4 = \frac{1}{sC_1}.$$

The transfer function for this second-order unity-gain low-pass filter is

$$H(s) = \frac{\overbrace{(2\pi f_c)^2}^{\omega_c^2}}{s^2 + \underbrace{2\pi \frac{f_c}{Q}}_{\frac{\omega_c}{Q} = 2\zeta\omega_c} s + \underbrace{(2\pi f_c)^2}_{\omega_c^2}}$$

where the cutoff frequency f_c and Q factor Q (i.e., damping ratio ζ) are given by

$$f_c = \frac{1}{2\pi\sqrt{R_1 R_2 C_1 C_2}}$$

and

$$2\zeta = \frac{1}{Q} = \frac{\sqrt{R_1 R_2 C_1 C_2}}{C_1}\left(\frac{1}{R_1} + \frac{1}{R_2}\right).$$

So,

$$Q = \frac{\sqrt{R_1 R_2 C_1 C_2}}{C_2\left(R_1 + R_2\right)}$$

The Q factor determines the height and width of the peak of the frequency response of the filter. As this parameter increases, the filter will tend to "ring" at a single resonant frequency near f_c.

A designer must choose the Q and f_c appropriate for his application. For example, a second-order Butterworth filter, which has maximally flat passband frequency response, has a Q of $1/\sqrt{2}$. Because there are two parameters and four unknowns, the design procedure typically fixes one resistor as a ratio of the other resistor and one capacitor as a ratio of the other capacitor. One possibility is to set the ratio between C_1 and C_2 as n and the ratio between R_1 and R_2 as m. So,

$$\begin{aligned} R_1 &= mR, \\ R_2 &= R, \\ C_1 &= nC, \\ C_2 &= C. \end{aligned}$$

Therefore, the f_c and Q expressions are

$$f_c = \frac{1}{2\pi\ RC\sqrt{mn}},$$

and

$$Q = \frac{\sqrt{mn}}{m+1}.$$

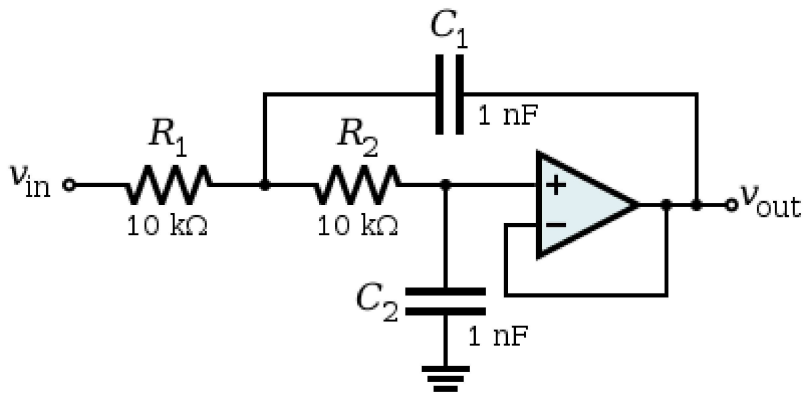


Figure 3: A low-pass filter, which is implemented with a Sallen–Key topology, with f_c=15.9 kHz and Q = 0.5.

For example, the circuit in Figure 3 has an f_c of $15.9\,\text{kHz}$ and a Q of 0.5. The transfer function is given by

$$H(s) = \frac{1}{1 + \underbrace{C_2(R_1 + R_2)}_{\frac{2\zeta}{\omega_c} = \frac{1}{\omega_c Q}} s + \underbrace{C_1 C_2 R_1 R_2}_{\frac{1}{\omega_c^2}} s^2},$$

and, after substitution, this expression is equal to

$$H(s) = \frac{1}{1 + \underbrace{RC(m+1)}_{\frac{2\zeta}{\omega_c} = \frac{1}{\omega_c Q}} s + \underbrace{mnR^2C^2}_{\frac{1}{\omega_c^2}} s^2}$$

which shows how every (R, C) combination comes with some (m, n) combination to provide the same f_c and Q for the low-pass filter. A similar design approach is used for the other filters below.

Example: High-pass filter

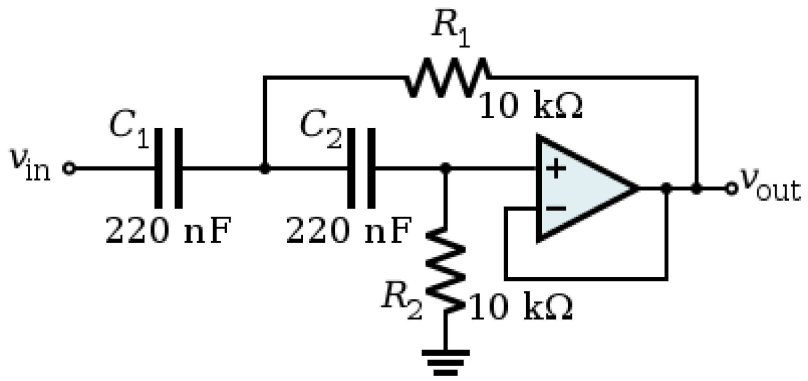


Figure 4: A specific Sallen–Key high-pass filter with f_c=72 Hz and Q = 0.5.

A second-order unity-gain high-pass filter with f_c of $72\ \mathrm{Hz}$ and Q of 0.5 is shown in Figure 4.

A second-order unity-gain high-pass filter has the transfer function

$$H(s) = \frac{s^2}{s^2 + \underbrace{2\pi\left(\frac{f_c}{Q}\right)}_{2\zeta\omega_c = \frac{\omega_c}{Q}} s + \underbrace{(2\pi f_c)^2}_{\omega_c^2}},$$

where cutoff frequency f_c and Q factor are discussed above in the low-pass filter discussion. The circuit above implements this transfer function by the equations

$$f_c = \frac{1}{2\pi\sqrt{R_1 R_2 C_1 C_2}}$$

(as before), and

$$\frac{1}{2\zeta} = Q = \frac{\sqrt{R_1 R_2 C_1 C_2}}{R_1 (C_1 + C_2)}.$$

So

$$2\zeta f_c = \frac{f_c}{Q} = \frac{C_1 + C_2}{2\pi R_2 C_1 C_2}.$$

Follow an approach similar to the one used to design the low-pass filter above.

VCVS Example: Bandpass configuration

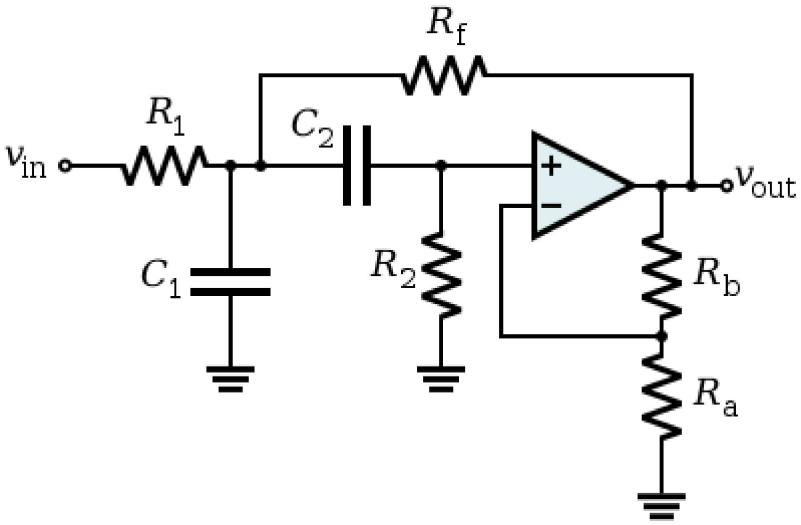


Figure 5: A bandpass filter realized with a VCVS topology.

An example of a non-unity-gain bandpass filter implemented with a VCVS filter is shown in Figure 5. Although it uses a different topology and an operational amplifier configured to provide non-unity-gain, it can be analyzed using similar methods as with the generic Sallen–Key topology. Its transfer function is given by:

$$H(s) = \frac{\overbrace{\left(1 + \frac{R_b}{R_a}\right)}^{G} \frac{s}{R_1 C_1}}{s^2 + \underbrace{\left(\frac{1}{R_1 C_1} + \frac{1}{R_2 C_1} + \frac{1}{R_2 C_2} - \frac{R_b}{R_a R_f C_1}\right)}_{2\zeta\omega_0 = \frac{\omega_0}{Q}} s + \underbrace{\frac{R_1 + R_f}{R_1 R_f R_2 C_1 C_2}}_{\omega_0^2 = (2\pi f_0)^2}}$$

The center frequency f_0 (i.e., the frequency where the magnitude response has its *peak*) is given by:

$$f_0 = \frac{1}{2\pi}\sqrt{\frac{R_f + R_1}{C_1 C_2 R_1 R_2 R_f}}$$

The voltage divider in the negative feedback loop controls the gain. The "inner gain" G provided by the operational amplifier is given by

$$G = 1 + \frac{R_b}{R_a}$$

while the amplifier gain at the peak frequency is given by:

$$A = \frac{G}{3 - G}$$

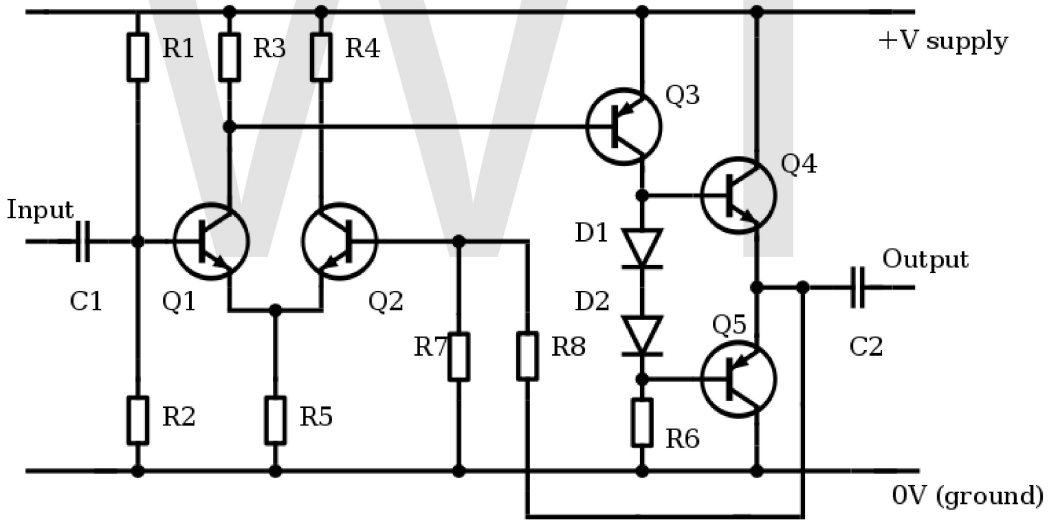
It can be seen that G must be kept below 3 or else the filter will oscillate. The filter is usually optimized by selecting $R_2 = 2R_1$ and $C_1 = C_2$.

Chapter 7

Introduction to Electronic Amplifier

The term **amplifier** as used here can mean either a circuit (or stage) using a single active device or a complete system such as a packaged audio hi-fi amplifier.

An **electronic amplifier** is a device for increasing the power of a signal. It does this by taking energy from a power supply and controlling the output to match the input signal shape but with a larger amplitude. In this sense, an amplifier may be considered as modulating the output of the power supply.



A practical amplifier circuit

Types of amplifier

Amplifiers can be specified according to their input and output properties. They have some kind of gain, or multiplication factor relating the magnitude of the output signal to the input signal. The gain may be specified as the ratio of output voltage to input voltage (voltage gain), output power to input power (power gain), or some combination of current, voltage and power. In many cases, with input and output in the same units, gain will be unitless (although often expressed in decibels); for others this is not necessarily

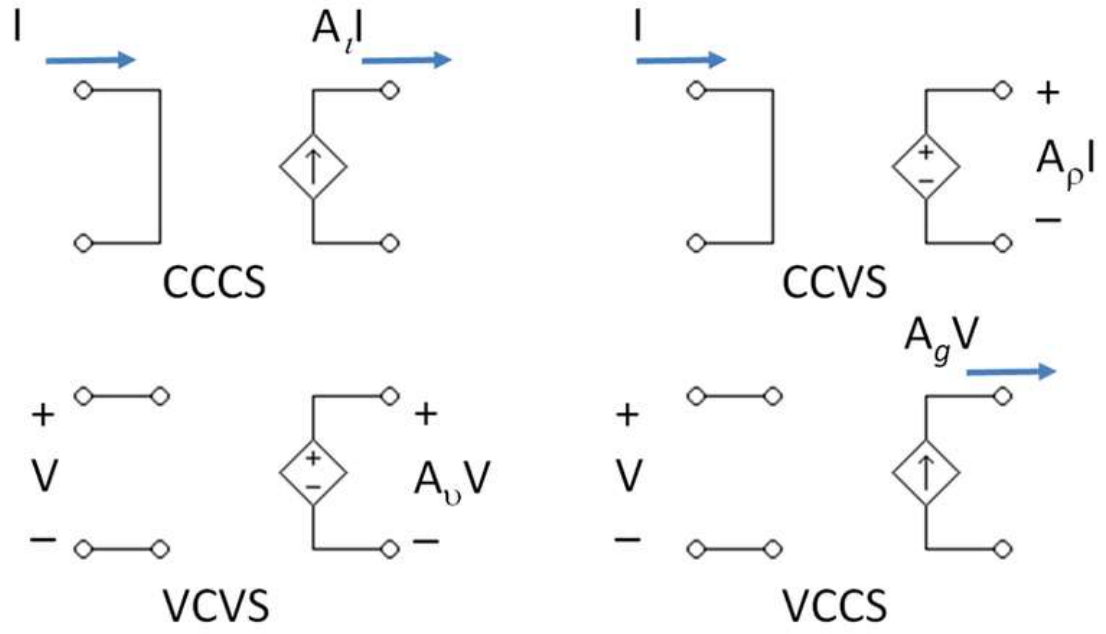
so. For example, a transconductance amplifier has a gain with units of conductance (output current per input voltage). The power gain of an amplifier depends on the source and load impedances used as well as its voltage gain; while an RF amplifier may have its impedances optimized for power transfer, audio and instrumentation amplifiers are normally employed with amplifier input and output impedances optimized for least loading and highest quality. So an amplifier that is said to have a gain of 20 dB might have a voltage gain of ten times and an available power gain of much more than 20 dB (100 times power ratio), yet be delivering a much lower power gain if, for example, the input is a 600 ohm microphone and the output is a 47 kilohm power amplifier's input socket.

In most cases an amplifier should be linear; that is, the gain should be constant for any combination of input and output signal. If the gain is not constant, e.g., by clipping the output signal at the limits of its capabilities, the output signal will be distorted. There are however cases where variable gain is useful.

Classification of amplifier stages and systems

There are many alternative classifications that address different aspects of amplifier designs, and they all express some particular perspective relating the design parameters to the objectives of the circuit. Amplifier design is always a compromise of numerous factors, such as cost, power consumption, real-world device imperfections, and a multitude of performance specifications. Below are several different approaches to classification:

Input and output variables



The four types of dependent source; control variable on left, output variable on right

Electronic amplifiers use two variables: current and voltage. Either can be used as input, and either as output leading to four types of amplifiers. In idealized form they are represented by each of the four types of dependent source used in linear analysis, as shown in the figure, namely:

Input	Output	Dependent source	Amplifier type
I	**I**	current controlled current source **CCCS**	current amplifier
I	**V**	current controlled voltage source **CCVS**	transresistance amplifier
V	**I**	voltage controlled current source **VCCS**	transconductance amplifier
V	**V**	voltage controlled voltage source **VCVS**	voltage amplifier

Each type of amplifier in its ideal form has an ideal input and output resistance that is the same as that of the corresponding dependent source:

Amplifier type	Dependent source	Input impedance	Output impedance
Current	**CCCS**	**0**	∞
Transresistance	**CCVS**	**0**	**0**
Transconductance	**VCCS**	∞	∞
Voltage	**VCVS**	∞	**0**

In practice the ideal impedances are only approximated. For any particular circuit, a small-signal analysis is often used to find the impedance actually achieved. A small-signal AC test current I_x is applied to the input or output node, all external sources are set to AC zero, and the corresponding alternating voltage V_x across the test current source determines the impedance seen at that node as $R = V_x / I_x$.

Amplifiers designed to attach to a transmission line at input and/or output, especially RF amplifiers, do not fit into this classification approach. Rather than dealing with voltage or current individually, they ideally couple with an input and/or output impedance matched to the transmission line impedance, that is, match *ratios* of voltage to current. Many real RF amplifiers come close to this ideal. Although, for a given appropriate source and load impedance, RF amplifiers can be characterized as amplifying voltage or current, they fundamentally are amplifying power.

Common terminal

One set of classifications for amplifiers is based on which device terminal is common to both the input and the output circuit. In the case of bipolar junction transistors, the three classes are common emitter, common base, and common collector. For field-effect transistors, the corresponding configurations are common source, common gate, and common drain; for triode vacuum devices, *common cathode*, *common grid*, and *common plate*. The output voltage of a common plate amplifier is the same as the input (this arrangement is used as the input presents a high impedance and does not load the signal source, although it does not amplify the voltage), i.e., the output at the cathode follows

the input at the grid; consequently it was commonly called a *cathode follower*. By analogy the terms *emitter follower* and *source follower* are sometimes used.

Unilateral or bilateral

When an amplifier has an output that exhibits no feedback to its input side, it is called **unilateral**. The input impedance of a unilateral amplifier is independent of the load, and the output impedance is independent of the signal source impedance.

If feedback connects part of the output back to the input of the amplifier it is called a **bilateral** amplifier. The input impedance of a bilateral amplifier is dependent upon the load, and the output impedance is dependent upon the signal source impedance.

All amplifiers are bilateral to some degree; however they may often be modeled as unilateral under operating conditions where feedback is small enough to neglect for most purposes, simplifying analysis.

Negative feedback is often applied deliberately to tailor amplifier behavior. Some feedback, which may be positive or negative, is unavoidable and often undesirable, introduced, for example, by parasitic elements such as the inherent capacitance between input and output of a device such as a transistor and capacitative coupling due to external wiring. Excessive frequency-dependent positive feedback may cause what is supposed to be an amplifier to become an oscillator.

Linear unilateral and bilateral amplifiers can be represented as two-port networks.

Inverting or non-inverting

Another way to classify amps is the phase relationship of the input signal to the output signal. An **inverting** amplifier produces an output 180 degrees out of phase with the input signal (that is, a polarity inversion or mirror image of the input as seen on an oscilloscope). A **non-inverting** amplifier maintains the phase of the input signal waveforms. An **emitter follower** is a type of non-inverting amplifier, indicating that the signal at the emitter of a transistor is following (that is, matching with unity gain but perhaps an offset) the input signal.

This description can apply to a single stage of an amplifier, or to a complete amplifier system.

Function

Other amplifiers may be classified by their function or output characteristics. These functional descriptions usually apply to complete amplifier systems or sub-systems and rarely to individual stages.

- A **servo amplifier** indicates an integrated feedback loop to actively control the output at some desired level. A **DC servo** indicates use at frequencies down to DC levels, where the rapid fluctuations of an audio or RF signal do not occur. These are often used in mechanical actuators, or devices such as DC motors that must maintain a constant speed or torque. An **AC servo** amp can do this for some ac motors.
- A **linear** amplifier responds to different frequency components independently, and does not generate harmonic distortion or Intermodulation distortion (well, hardly any). A **nonlinear** amplifier does generate distortion (e.g. the output is a current to a lamp that must be either fully on or off, but the input is continuously variable; or the amplifier is used in an analog computer where a special transfer function, such as logarithmic, is desired; or a following tuned circuit will remove the harmonics generated by a non-linear RF amplifier).
- A **wideband** amplifier has a precise amplification factor over a wide range of frequencies, and is often used to boost signals for relay in communications systems. A **narrowband** amp is made to amplify only a specific narrow range of frequencies, to the exclusion of other frequencies.
- An **RF** amplifier refers to an amplifier designed for use in the radio frequency range of the electromagnetic spectrum, and is often used to increase the sensitivity of a receiver or the output power of a transmitter.
- An **audio amplifier** is designed for use in reproducing audio frequencies. This category subdivides into small signal amplification, and power amps which are optimised for driving speakers, sometimes with multiple amps grouped together as separate or bridgeable channels to accommodate different audio reproduction requirements. Frequently used terms within audio amplifiers include:
    - preamplifier (preamp), that may include phono or gramophone preamp with equalization for RIAA LP recordings, or tape head preamps with CCIR equalisation filters; they may include filters or tone control circuitry.
    - power amplifier (normally assumed to drive loudspeakers), headphone amplifiers, and public address amplifiers.
    - stereo amplifiers imply two channels of output (left and right), although the term simply means "solid" sound (referring to three-dimensional) - so quadraphonic stereo was used for amplifiers with 4 channels; 5.1 and 7.1 systems refer to Home theatre systems with 5 or 7 normal spacial channels, plus a subwoofer channel (that is not very directional).
- Buffer amplifiers, that may include emitter followers, provide a high impedance input for a device (perhaps another amplifier, or perhaps an energy-hungry load such as lights) that would otherwise draw too much current from the source. Line drivers are a type of buffer intended to feed long or interference-prone interconnect cables, possibly with differential outputs if driving twisted pairs of cables.
- A special type of amplifier is widely used in instruments and for signal processing, among many other varied uses. These are known as **operational amplifiers** or **op-amps**. This is because this type of amplifier is used in circuits that perform mathematical algorithmic functions, or "operations" on input signals

to obtain specific types of output signals. A typical modern op-amp has differential inputs (one "inverting", one "non-inverting") and one output. An idealised op-amp has the following characteristics:

- Infinite input impedance (so as to not load circuitry it is sampling as a control input)
- Zero output impedance
- Infinite gain
- Zero propagation delay

The performance of an op-amp with these characteristics would be entirely defined by the (usually passive) components forming a negative feedback loop around it, that is, *the amplifier itself has no effect on the output*.
Today, op-amps are usually provided as integrated circuits, rather than constructed from discrete components. All real-world op-amps fall short of the idealised specification above – but some modern components have remarkable performance and come close in some respects.

Interstage coupling method

Amplifiers are sometimes classified by the coupling method of the signal at the input, output, or between stages. Different types of these include:

Resistive-capacitive (RC) coupled amplifier, using a network of resistors and capacitors
: By design these amplifiers cannot amplify DC signals as the capacitors block the DC component of the input signal. RC-coupled amplifiers were used very often in circuits with vacuum tubes or discrete transistors. In the days of the integrated circuit a few more transistors on a chip are much cheaper and smaller than a capacitor.

Inductive-capacitive (LC) coupled amplifier, using a network of inductors and capacitors
: This kind of amplifier is most often used in selective radio-frequency circuits.

Transformer coupled amplifier, using a transformer to match impedances or to decouple parts of the circuits
: Quite often LC-coupled and transformer-coupled amplifiers cannot be distinguished as a transformer is some kind of inductor.

Direct coupled amplifier, using no impedance and bias matching components
: This class of amplifier was very uncommon in the vacuum tube days when the anode (output) voltage was at greater than several hundred volts and the grid (input) voltage at a few volts minus. So they were only used if the gain was specified down to DC (e.g., in an oscilloscope). In the context of modern electronics developers are encouraged to use directly coupled amplifiers whenever possible.

Frequency range

Depending on the frequency range and other properties amplifiers are designed according to different principles.

- Frequency ranges down to DC are only used when this property is needed. DC amplification leads to specific complications that are avoided if possible; **DC-blocking** capacitors are added to remove DC and sub-sonic frequencies from audio amplifiers.
- Depending on the frequency range specified different design principles must be used. Up to the MHz range only "discrete" properties need be considered; e.g., a terminal has an input impedance.
- As soon as any connection within the circuit gets longer than perhaps 1% of the wavelength of the highest specified frequency (e.g., at 100 MHz the wavelength is 3 m, so the critical connection length is approx. 3 cm) design properties radically change. For example, a specified length and width of a PCB trace can be used as a selective or impedance-matching entity.
- Above a few 100 MHz, it gets difficult to use discrete elements, especially inductors. In most cases PCB traces of very closely defined shapes are used instead.

The frequency range handled by an amplifier might be specified in terms of bandwidth (normally implying a response that is 3 dB down when the frequency reaches the specified bandwidth), or by specifying a frequency response that is within a certain number of deciBels between a lower and an upper frequency (e.g. "20 Hz to 20 kHz plus or minus 1 dB").

Type of load

- Untuned
  - audio
  - video
- Tuned (RF amps) - used for amplifying a single radio frequency or a band of frequencies

Implementation

Amplifiers are implemented using active elements of different kinds:

- The first active elements were relays. They were for example used in trans-continental telegraph lines: a weak current was used to switch the voltage of a battery to the outgoing line.
- For transmitting audio, carbon microphones were used as the active element. This was used to modulate a radio-frequency source in one of the first AM audio transmissions, by Reginald Fessenden on Dec. 24, 1906.
- In the 1960s, the transistor started to take over. These days, discrete transistors are still used in high-power amplifiers and in specialist audio devices.
- Up to the early 1970s, most amplifiers used vacuum tubes ("valves" in the UK). Today, tubes are used for specialist audio applications such as guitar amplifiers and audiophile amplifiers. Many broadcast transmitters still use vacuum tubes.

- Beginning in the 1970s, more and more transistors were connected on a single chip therefore creating the integrated circuit. Nearly all amplifiers commercially available today are based on integrated circuits.

For exotic purposes, other active elements have been used. For example, in the early days of the communication satellite parametric amplifiers were used. The core circuit was a diode whose capacity was changed by an RF signal created locally. Under certain conditions, this RF signal provided energy that was modulated by the extremely weak satellite signal received at the earth station. The operating principle of a parametric amplifier is somewhat similar to the principle by which children keep their swings in motion: as long as the swing moves you only need to change a parameter of the swinging entity; e.g., you must move your center of gravity up and down. In our case, the capacity of the diode is changed periodically.

Power amplifier classes

Angle of flow or conduction angle

Power amplifier circuits (output stages) are classified as A, B, AB and C for analog designs, and class D and E for switching designs based upon the conduction angle or *angle of flow*, Θ, of the input signal through the (or each) output amplifying device, that is, the portion of the input signal cycle during which the amplifying device conducts. The image of the conduction angle is derived from amplifying a sinusoidal signal. (If the device is always on, Θ = 360°.) The angle of flow is closely related to the amplifier power efficiency. The various classes are introduced below, followed by more detailed discussion under individual headings later on.

Class A

100% of the input signal is used (conduction angle Θ = 360° or 2π); i.e., the active element remains conducting (works in its "linear" range) all of the time. Where efficiency is not a consideration, most small signal linear amplifiers are designed as Class A. Class A amplifiers are typically more linear and less complex than other types, but are very inefficient. This type of amplifier is most commonly used in small-signal stages or for low-power applications (such as driving headphones). Subclass A2 is sometimes used to refer to vacuum tube Class A stages where the grid is allowed to be driven slightly positive on signal peaks, resulting in slightly more power than normal Class A (A1; where the grid is always negative), but incurring more distortion.

Class B

50% of the input signal is used (Θ = 180° or π; i.e., the active element works in its linear range half of the time and is more or less turned off for the other half). In most Class B, there are two output devices (or sets of output devices), each of which conducts alternately (push–pull) for exactly 180° (or half cycle) of the input signal; selective RF amplifiers can also be implemented using a single active element.

These amplifiers are subject to *crossover distortion* if the transition from one active element to the other is not perfect, as when two complementary transistors (i.e., one PNP, one NPN) are connected as two emitter followers with their base and emitter terminals in common, requiring the base voltage to slew across the region where both devices are turned off.

Class AB

Here the two active elements conduct more than half of the time as a means to reduce the cross-over distortions of Class B amplifiers. In the example of the complementary emitter followers a bias network allows for more or less quiescent current thus providing an operating point somewhere between Class A and Class B. Sometimes a figure is added (e.g., AB_1 or AB_2) for vacuum tube stages where the grid voltage is always negative with respect to the cathode (Class AB_1) or may be slightly positive (hence drawing grid current, adding more distortion, but giving slightly higher output power) on signal peaks (Class AB_2); another interpretation being higher figures implying a higher quiescent current and therefore more of the properties of Class A.

Class C

Less than 50% of the input signal is used (conduction angle $\Theta < 180°$). The advantage is potentially high efficiency, but a disadvantage is high distortion.

Class D

These use switching to achieve a very high power efficiency (more than 90% in modern designs). By allowing each output device to be either fully on or off, losses are minimized. The analog output is created by pulse-width modulation; i.e., the active element is switched on for shorter or longer intervals instead of modifying its resistance. There are more complicated switching schemes like sigma-delta modulation, to improve some performance aspects like lower distortions or better efficiency.

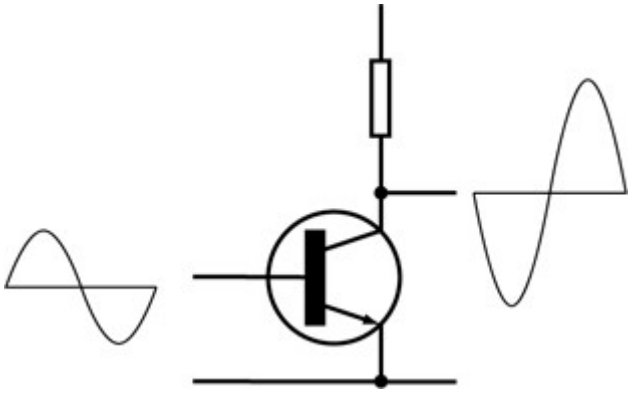
Additional classes

There are several other amplifier classes, although they are mainly variations of the previous classes. For example, Class G and Class H amplifiers are marked by variation of the supply rails (in discrete steps or in a continuous fashion, respectively) following the input signal. Wasted heat on the output devices can be reduced as excess voltage is kept to a minimum. The amplifier that is fed with these rails itself can be of any class. These kinds of amplifiers are more complex, and are mainly used for specialized applications, such as very high-power units. Also, Class E and Class F amplifiers are commonly described in literature for radio frequencies applications where efficiency of the traditional classes in are important, yet several aspects not covered elsewhere (e.g.: amplifiers often simply said to have a gain of x dB - so what power gain?) deviate substantially from their ideal values. These classes use harmonic tuning of their output networks to achieve higher efficiency and can be considered a subset of Class C due to their conduction angle characteristics.

The classes can be most easily understood using the diagrams in each section below. For the sake of illustration, a bipolar junction transistor is shown as the amplifying device,

but in practice this could be a MOSFET or vacuum tube device. In an analog amplifier (the most common kind), the signal is applied to the input terminal of the device (base, gate or grid), and this causes a proportional output drive current to flow out of the output terminal. The output drive current comes from the power supply.

Class A



Class A amplifier

Amplifying devices operating in Class A conduct over the whole of the input cycle such that the output signal is an exact scaled-up replica of the input with no clipping. A *Class A amplifier* (or operational amplifier) is distinguished by the *output stage* (and perhaps the driver) device(s) being biased into Class A; even Class AB and B amplifiers normally have early stages operating in Class A. Class A is the usual means of implementing small-signal amplifiers, so the term *Class A design* applied to equipment such as preamplifiers (for example, in recording studios) implies not so much their *use* of Class A, but that their sound is top quality - good enough to be matched with top quality Class A power amplifiers.

Advantages of Class A Amplifiers

- Class A designs are simpler than other classes; for example Class AB and B designs require two devices (push-pull output) to handle both halves of the waveform, and circuitry to keep the quiescent bias optimal during temperature changes; Class A can use either single-ended or push-pull and bias is usually less critical.
- The amplifying element is biased so the device is always conducting to some extent, normally implying the quiescent (small-signal) collector current (for transistors; drain current for FETs or anode/plate current for vacuum tubes) is close to the most linear portion (sometimes called the "sweet spot") of its characteristic curve (known as its transfer characteristic or transconductance curve), giving the least audio distortion.
- Because the device is never shut off completely there is no "turn on" time, little problem with charge storage, and generally better high frequency performance and feedback loop stability (and usually fewer high-order harmonics).

- The point at which the device comes closest to being cut off (and so significant change in gain, hence non-linearity) is not close to zero signal, so the problem of crossover distortion associated with Class AB and B designs is avoided, even in Class A double-ended stages.

Disadvantage of Class A Amplifiers

- They are very inefficient; a theoretical maximum of 50% is obtainable with inductive output coupling and only 25% with capacitive coupling, unless Square law output stages are used. In a power amplifier this not only wastes power and limits battery operation, it may place restrictions on the output devices that can be used (for example: ruling out some audio triodes if modern low-efficiency loudspeakers are to be used), and will increase costs. Inefficiency comes not just from the fact that the device is always conducting to some extent (that happens even with Class AB, yet its efficiency can be close to that of Class B); it is that the standing current is roughly half the maximum output current (although this can be less with Square law output stage), together with the problem that a large part of the power supply voltage is developed across the output device at low signal levels (as with Classes AB and B, but unlike output stages such as Class D). If high output powers are needed from a Class A circuit, the power waste (and the accompanying heat) will become significant. For every watt delivered to the load, the amplifier itself will, *at best*, dissipate another watt. For large powers this means very large and expensive power supplies and heat sinking.

Class A designs have largely been superseded by the more efficient designs for power amplifiers, though they remain popular with some hobbyists, mostly for their simplicity. Also, many audiophiles believe that Class A gives the best sound quality (for their absence of crossover distortion and reduced odd-harmonic and high-order harmonic distortion) which provides a small market for expensive **high fidelity** Class A amps.

Single-Ended and Triode Class A Amplifiers

Some aficionados who prefer Class A amplifiers also prefer the use of thermionic valve (or "tube") designs instead of transistors, especially in Single-ended triode output configurations for several claimed reasons:

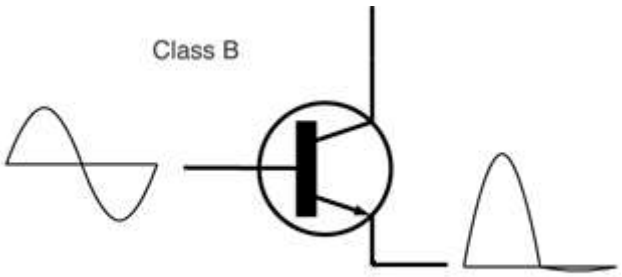
- Single-ended output stages (be they tube or transistor) have an asymmetrical transfer function, meaning that even harmonics in the created distortion tend not to be canceled (as they are in push-pull output stages); by using tubes OR FETs most of the distortion is from the square law transfer characteristic and so second-order, which some consider to be "warmer" and more pleasant.
- For those who prefer low distortion figures, the use of tubes with Class A (generating little odd-harmonic distortion, as mentioned above) together with symmetrical circuits (such as push-pull output stages, or balanced low-level stages) results in the cancellation of most of the even distortion harmonics, hence the removal of most of the distortion.

- Though good amplifier design can reduce harmonic distortion patterns to almost nothing, distortion is essential to the sound of electric guitar amplifiers, for example, and is held by recording engineers to offer more flattering microphones and to enhance "clinical-sounding" digital technology.
- Historically, valve amplifiers often used a Class A power amplifier simply because valves are large and expensive; many Class A designs use only a single device.

Transistors are much cheaper, and so more elaborate designs that give greater efficiency but use more parts are still cost-effective. A classic application for a pair of class A devices is the long-tailed pair, which is exceptionally linear, and forms the basis of many more complex circuits, including many audio amplifiers and almost all op-amps. Class A amplifiers are often used in output stages of high quality op-amps (although the accuracy of the bias in low cost op-amps such as the **741** may result in Class A or Class AB or Class B, varying from device to device or with temperature). They are sometimes used as medium-power, low-efficiency, and high-cost audio amplifiers. The power consumption is unrelated to the output power. At idle (no input), the power consumption is essentially the same as at high output volume. The result is low efficiency and high heat dissipation.

Class B and AB

Class B amplifiers only amplify half of the input wave cycle, thus creating a large amount of distortion, but their efficiency is greatly improved and is much better than Class A. Class B has a maximum theoretical efficiency of 78.5% (i.e., $\pi/4$). This is because the amplifying element is switched off altogether half of the time, and so cannot dissipate power. A single Class B element is rarely found in practice, though it has been used for driving the loudspeaker in the early IBM Personal Computers with beeps, and it can be used in RF power amplifier where the distortion levels are less important. However Class C is more commonly used for this.



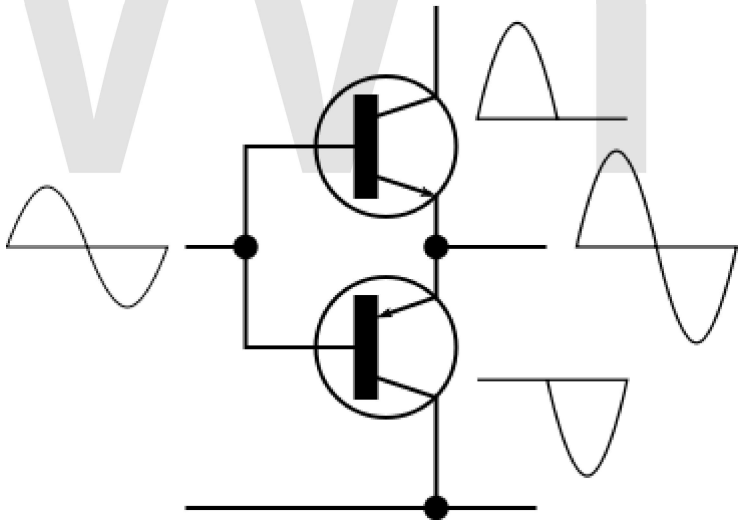
Class B Amplifier

A practical circuit using Class B elements is the push-pull stage, such as the very simplified complementary pair arrangement shown below. Here, complementary or quasi-complementary devices are each used for amplifying the opposite halves of the input signal, which is then recombined at the output. This arrangement gives excellent efficiency, but can suffer from the drawback that there is a small mismatch in the cross-

over region - at the "joins" between the two halves of the signal, as one output device has to take over supplying power exactly as the other finishes. This is called crossover distortion. An improvement is to bias the devices so they are not completely off when they're not in use. This approach is called *Class AB* operation.

In Class AB operation, each device operates the same way as in Class B over half the waveform, but also conducts a small amount on the other half. As a result, the region where both devices simultaneously are nearly off (the "dead zone") is reduced. The result is that when the waveforms from the two devices are combined, the crossover is greatly minimised or eliminated altogether. The exact choice of **quiescent current**, the standing current through both devices when there is no signal, makes a large difference to the level of distortion (and to the risk of thermal runaway, that may damage the devices); often the bias voltage applied to set this quiescent current has to be adjusted with the temperature of the output transistors (for example in the circuit at the beginning, the diodes would be mounted physically close to the output transistors, and chosen to have a matched temperature coefficient). Another approach (often used as well as thermally-tracking bias voltages) is to include small value resistors in series with the emitters.

Class AB sacrifices some efficiency over class B in favor of linearity, thus is less efficient (below 78.5% for full-amplitude sinewaves in transistor amplifiers, typically; much less is common in Class AB vacuum tube amplifiers). It is typically much more efficient than class A.



Class B push–pull amplifier

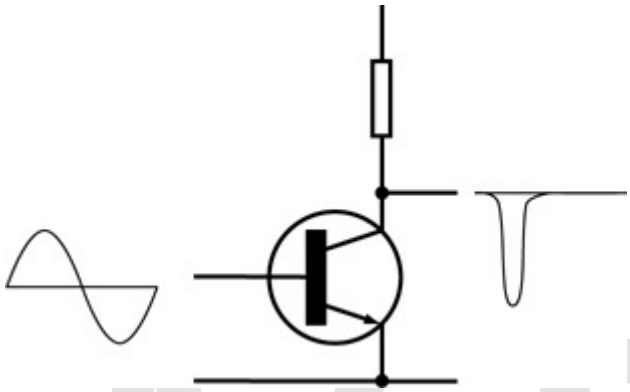
Class B or AB push–pull circuits are the most common design type found in audio power amplifiers. Class AB is widely considered a good compromise for audio amplifiers, since much of the time the music is quiet enough that the signal stays in the "class A" region, where it is amplified with good fidelity, and by definition if passing out of this region, is large enough that the distortion products typical of class B are relatively small. The crossover distortion can be reduced further by using negative feedback. Class B and AB

amplifiers are sometimes used for RF linear amplifiers as well. Class B amplifiers are also favored in battery-operated devices, such as transistor radios.

Digital Class B

A limited power output Class-B amplifier with a single-ended supply rail of 5±0.5 V.

Class C



Class C amplifier

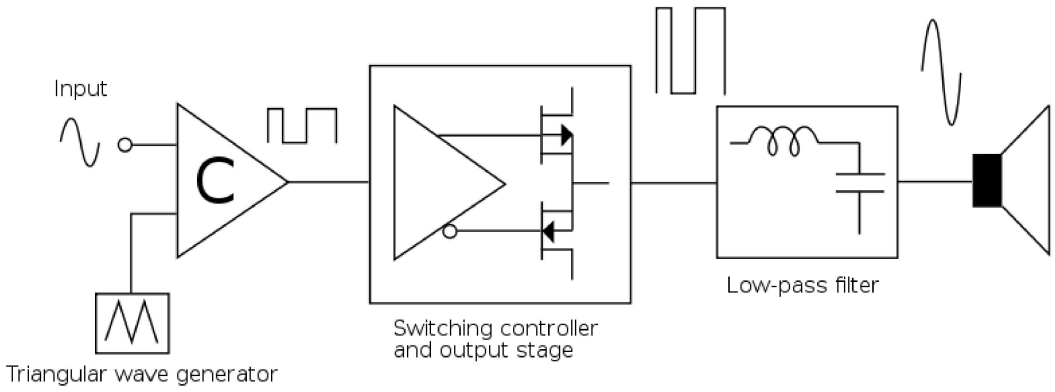
Class C amplifiers conduct less than 50% of the input signal and the distortion at the output is high, but high efficiencies (up to 90%) are possible. Some applications (for example, megaphones) can tolerate the distortion. A much more common application for Class C amplifiers is in RF transmitters, where the distortion can be vastly reduced by using tuned loads on the amplifier stage. The input signal is used to roughly switch the amplifying device on and off, which causes pulses of current to flow through a tuned circuit.

The Class C amplifier has two modes of operation: tuned and untuned. The diagram shows a waveform from a simple class C circuit without the tuned load. This is called untuned operation, and the analysis of the waveforms shows the massive distortion that appears in the signal. When the proper load (e.g., a pure inductive-capacitive filter) is used, two things happen. The first is that the output's bias level is clamped, so that the output variation is centered at one-half of the supply voltage. This is why tuned operation is sometimes called a *clamper*. This action of elevating bias level allows the waveform to be restored to its proper shape, allowing a complete waveform to be re-established despite having only a one-polarity supply. This is directly related to the second phenomenon: the waveform on the center frequency becomes much less distorted. The distortion that is present is dependent upon the bandwidth of the tuned load, with the center frequency seeing very little distortion, but greater attenuation the farther from the tuned frequency that the signal gets.

The tuned circuit will only resonate at particular frequencies, and so the unwanted frequencies are dramatically suppressed, and the wanted full signal (sine wave) will be extracted by the tuned load (e.g., a high-quality bell will ring at a particular frequency

when it is hit periodically with a hammer). Provided the transmitter is not required to operate over a very wide band of frequencies, this arrangement works extremely well. Other residual harmonics can be removed using a filter.

Class D



Block diagram of a basic switching or PWM (Class-D) amplifier

Boss Audio Class D mono car audio amplifier with a low pass filter for powering subwoofers

In the Class D amplifier the input signal is converted to a sequence of higher voltage output pulses. The averaged-over-time power values of these pulses are directly proportional to the instantaneous amplitude of the input signal. The frequency of the output pulses is typically ten or more times the highest frequency in the input signal to be amplified. The output pulses contain inaccurate spectral components (that is, the pulse frequency and its harmonics) which must be removed by a low-pass passive filter. The resulting filtered signal is then an amplified replica of the input.

These amplifiers use pulse width modulation, pulse density modulation (sometimes referred to as pulse frequency modulation) or more advanced form of modulation such as Delta-sigma modulation (for example, in the Analog Devices AD1990 Class-D audio power amplifier). Output stages such as those used in pulse generators are examples of Class D amplifiers. The term Class D is usually applied to devices intended to reproduce signals with a bandwidth well below the switching frequency.

Class D amplifiers can be controlled by either analog or digital circuits. The digital control introduces additional distortion called *quantization error* caused by its conversion of the input signal to a digital value.

The main advantage of a Class D amplifier is power efficiency. Because the output pulses have a fixed amplitude, the switching elements (usually MOSFETs, but valves and bipolar transistors were once used) are switched either completely on or completely off, rather than operated in linear mode. A MOSFET operates with the lowest resistance when fully-on and thus has the lowest power dissipation when in that condition, except when fully off. When operated in a linear mode the MOSFET has variable amounts of resistance that vary linearly with the input voltage and the resistance is something other than the minimum possible, therefore more electrical energy is dissipated as heat. Compared to Class A/B operation, Class D's lower losses permit the use of a smaller heat sink for the MOSFETS while also reducing the amount of AC power supply power required. Thus, Class D amplifiers do not need as large or as heavy power supply transformers or heatsinks, so they are smaller and more compact in size than an equivalent Class AB amplifier.

Class D amplifiers have been widely used to control motors, and almost exclusively for small DC motors, but they are now also used as audio amplifiers, with some extra circuitry to allow analogue to be converted to a much higher frequency pulse width modulated signal. The relative difficulty of achieving good audio quality means that nearly all are used in applications where quality is not a factor, such as modestly-priced bookshelf audio systems and "DVD-receivers" in mid-price home theater systems.

High quality Class D audio amplifiers are now, however, starting to appear in the market:

- Tripath have called their revised Class D designs Class T.
- Bang and Olufsen's ICEPower Class D system has been used in the Alpine PDX range and some of Pioneer's PRS range and for other manufacturers' equipment.

These revised designs have been said to rival good traditional AB amplifiers in terms of quality.

Before these higher quality designs existed an earlier use of Class D amplifiers and prolific area of application was high-powered, subwoofer amplifiers in cars. Because subwoofers are generally limited to a bandwidth of no higher than 150 Hz, the switching speed for the amplifier does not have to be as high as for a full range amplifier. The drawback with Class D designs being used to power subwoofers is that their output filters (typically inductors that convert the pulse width signal back into an analogue waveform) lower the damping factor of the amplifier.

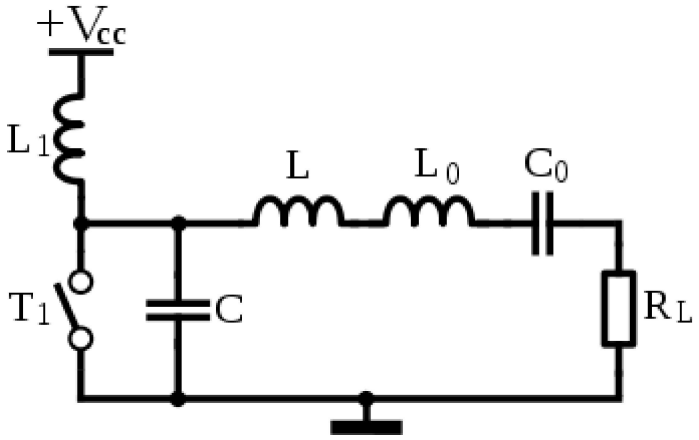
This means that the amplifier cannot prevent the subwoofer's reactive nature from lessening the impact of low bass sounds (as explained in the feedback part of the Class AB section). Class D amplifiers for driving subwoofers are relatively inexpensive, in comparison to Class AB amplifiers. A 1000 W Class D subwoofer amplifier that can operate at about 80% to 95% efficiency costs about US$250, much less than a Class AB amplifier of this power, which would cost several thousand dollars.

The letter *D* used to designate this amplifier class is simply the next letter after *C*, and does not stand for *digital*. Class D and Class E amplifiers are sometimes mistakenly described as "digital" because the output waveform superficially resembles a pulse-train of digital symbols, but a Class D amplifier merely converts an input waveform into a continuously pulse-width modulated (square wave) analog signal. (A digital waveform would be pulse-code modulated.)

Additional classes

Class E

The Class E/F amplifier is a highly efficient switching power amplifier, typically used at such high frequencies that the switching time becomes comparable to the duty time. As said in the Class D amplifier, the transistor is connected via a serial LC circuit to the load, and connected via a large L (inductor) to the supply voltage. The supply voltage is connected to ground via a large capacitor to prevent any RF signals leaking into the supply. The Class E amplifier adds a C (capacitor) between the transistor and ground and uses a defined L_1 to connect to the supply voltage.



Class E Amplifier

The following description ignores DC, which can be added easily afterwards. The above mentioned C and L are in effect a parallel LC circuit to ground. When the transistor is on, it pushes through the serial LC circuit into the load and some current begins to flow to the parallel LC circuit to ground. Then the serial LC circuit swings back and compensates the current into the parallel LC circuit. At this point the current through the transistor is zero and it is switched off. Both LC circuits are now filled with energy in C and L_0. The whole circuit performs a damped oscillation. The damping by the load has been adjusted so that some time later the energy from the Ls is gone into the load, but the energy in both C_0 peaks at the original value to in turn restore the original voltage so that the voltage across the transistor is zero again and it can be switched on.

With load, frequency, and duty cycle (0.5) as given parameters and the constraint that the voltage is not only restored, but peaks at the original voltage, the four parameters (L, L_0, C and C_0) are determined. The Class E amplifier takes the finite on resistance into account and tries to make the current touch the bottom at zero. This means that the voltage and the current at the transistor are symmetric with respect to time. The Fourier transform allows an elegant formulation to generate the complicated LC networks and says that the first harmonic is passed into the load, all even harmonics are shorted and all higher odd harmonics are open.

Class E uses a significant amount of second-harmonic voltage. The second harmonic can be used to reduce the overlap with edges with finite sharpness. For this to work, energy on the second harmonic has to flow from the load into the transistor, and no source for this is visible in the circuit diagram. In reality, the impedance is mostly reactive and the only reason for it is that Class E is a Class F (see below) amplifier with a much simplified load network and thus has to deal with imperfections.

In many amateur simulations of Class E amplifiers, sharp current edges are assumed nullifying the very motivation for Class E and measurements near the transit frequency of the transistors show very symmetric curves, which look much similar to Class F simulations.

The Class E amplifier was invented in 1972 by Nathan O. Sokal and Alan D. Sokal, and details were first published in 1975. Some earlier reports on this operating class have been published in Russian.

Class F

In push-pull amplifiers and in CMOS, the even harmonics of both transistors just cancel. Experiment shows that a square wave can be generated by those amplifiers and theory shows that square waves do consist of odd harmonics only. In a Class D amplifier, the output filter blocks all harmonics; i.e., the harmonics see an open load. So even small currents in the harmonics suffice to generate a voltage square wave. The current is in phase with the voltage applied to the filter, but the voltage across the transistors is out of phase. Therefore, there is a minimal overlap between current through the transistors and voltage across the transistors. The sharper the edges, the lower the overlap.

While Class D sees the transistors and the load as two separate modules, Class F admits imperfections like the parasitics of the transistor and tries to optimise the global system to have a high impedance at the harmonics. Of course there has to be a finite voltage across the transistor to push the current across the on-state resistance. Because the combined current through both transistors is mostly in the first harmonic, it looks like a sine. That means that in the middle of the square the maximum of current has to flow, so it may make sense to have a dip in the square or in other words to allow some overswing of the voltage square wave. A Class F load network by definition has to transmit below a cutoff frequency and reflect above.

Any frequency lying below the cutoff and having its second harmonic above the cutoff can be amplified, that is an octave bandwidth. On the other hand, an inductive-capacitive series circuit with a large inductance and a tunable capacitance may be simpler to implement. By reducing the duty cycle below 0.5, the output amplitude can be modulated. The voltage square waveform will degrade, but any overheating is compensated by the lower overall power flowing. Any load mismatch behind the filter can only act on the first harmonic current waveform, clearly only a purely resistive load makes sense, then the lower the resistance, the higher the current.

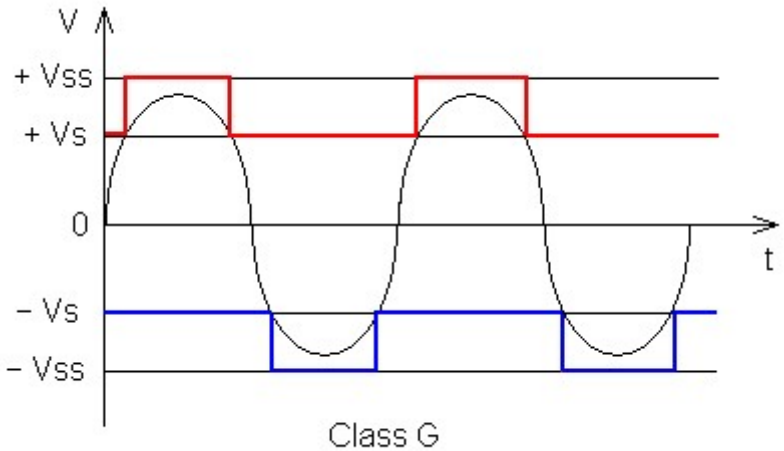
Class F can be driven by sine or by a square wave, for a sine the input can be tuned by an inductor to increase gain. If Class F is implemented with a single transistor, the filter is complicated to short the even harmonics. All previous designs use sharp edges to minimise the overlap.

Classes G and H

There are a variety of amplifier designs that enhance Class AB output stages with more efficient techniques to achieve greater efficiencies with low distortion. These designs are common in large audio amplifiers since the heatsinks and power transformers would be prohibitively large (and costly) without the efficiency increases. The terms "Class G" and

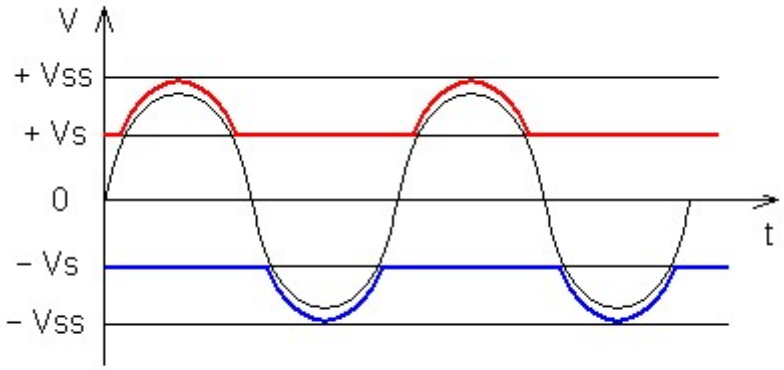
"Class H" are used interchangeably to refer to different designs, varying in definition from one manufacturer or paper to another.

Class G amplifiers (which use "rail switching" to decrease power consumption and increase efficiency) are more efficient than Class AB amplifiers. These amplifiers provide several power rails at different voltages and switch between them as the signal output approaches each level. Thus, the amplifier increases efficiency by reducing the wasted power at the output transistors. Class G amplifiers are more efficient than Class AB but less efficient when compared to Class D, without the negative EMI effects of Class D.



Class G

Class H amplifiers take the idea of Class G one step further creating an infinitely variable supply rail. This is done by modulating the supply rails so that the rails are only a few volts larger than the output signal at any given time. The output stage operates at its maximum efficiency all the time. Switched-mode power supplies can be used to create the tracking rails. Significant efficiency gains can be achieved but with the drawback of more complicated supply design and reduced THD performance.



Class H

The voltage signal shown is thus a larger version of the input, but has been changed in sign (inverted) by the amplification. Other arrangements of amplifying device are possible, but that given (that is, common emitter, common source or common cathode) is the easiest to understand and employ in practice. If the amplifying element is linear, then the output will be faithful copy of the input, only larger and inverted. In practice, transistors are not linear, and the output will only approximate the input. Non-linearity from any of several sources is the origin of distortion within an amplifier. Which class of amplifier (A, B, AB or C) depends on how the amplifying device is biased — in the diagrams the bias circuits are omitted for clarity.

Any real amplifier is an imperfect realization of an ideal amplifier. One important limitation of a real amplifier is that the output it can generate is ultimately limited by the power available from the power supply. An amplifier will saturate and clip the output if the input signal becomes too large for the amplifier to reproduce or if operational limits for a device are exceeded.

For additional information on Class H: Efficiency Class H

Doherty amplifiers

A hybrid configuration receiving new attention is the Doherty amplifier, invented in 1934 by William H. Doherty for Bell Laboratories (whose sister company, Western Electric, was then an important manufacturer of radio transmitters). The Doherty amplifier consists of a Class B *primary* or *carrier* stage in parallel with a Class C *auxiliary* or *peak* stage. The input signal is split to drive the two amplifiers and a combining network sums the two output signals. Phase shifting networks are employed in the inputs and the outputs. During periods of low signal level, the Class B amplifier efficiently operates on the signal and the Class C amplifier is cutoff and consumes little power. During periods of high signal level, the Class B amplifier delivers its maximum power and the Class C amplifier delivers up to its maximum power. The efficiency of previous AM transmitter designs was proportional to modulation but, with average modulation typically around 20%, transmitters were limited to less than 50% efficiency. In Doherty's design, even with zero modulation, a transmitter could achieve at least 60% efficiency.

As a successor to Western Electric for broadcast transmitters, the Doherty concept was considerably refined by Continental Electronics Manufacturing Company of Dallas, TX. Perhaps, the ultimate refinement was the screen-grid modulation scheme invented by Joseph B. Sainton. The Sainton amplifier consists of a Class C primary or carrier stage in parallel with a Class C auxiliary or peak stage. The stages are split and combined through 90-degree phase shifting networks as in the Doherty amplifier. The unmodulated radio frequency carrier is applied to the control grids of both tubes. Carrier modulation is applied to the screen grids of both tubes. The bias point of the carrier and peak tubes is different, and is established such that the peak tube is cutoff when modulation is absent (and the amplifier is producing rated unmodulated carrier power) whereas both tubes contribute twice the rated carrier power during 100% modulation (as four times the carrier power is required to achieve 100% modulation). As both tubes operate in Class C,

a significant improvement in efficiency is thereby achieved in the final stage. In addition, as the tetrode carrier and peak tubes require very little drive power, a significant improvement in efficiency within the driver stage is achieved as well (317C, et al.). The released version of the Sainton amplifier employs a cathode-follower modulator, not a push-pull modulator. Previous Continental Electronics designs, by James O. Weldon and others, retained most of the characteristics of the Doherty amplifier but added screen-grid modulation of the driver (317B, et al.).

The Doherty amplifier remains in use in very-high-power AM transmitters, but for lower-power AM transmitters, vacuum-tube amplifiers in general were eclipsed in the 1980s by arrays of solid-state amplifiers, which could be switched on and off with much finer granularity in response to the requirements of the input audio. However, interest in the Doherty configuration has been revived by cellular-telephone and wireless-Internet applications where the sum of several constant-envelope users creates an aggregate AM result. The main challenge of the Doherty amplifier for digital transmission modes is in aligning the two stages and getting the class-C amplifier to turn on and off very quickly.

Recently, Doherty amplifiers have found widespread use in cellular base station transmitters for GHz frequencies. Implementations for transmitters in mobile devices have also been demonstrated.

Special classes

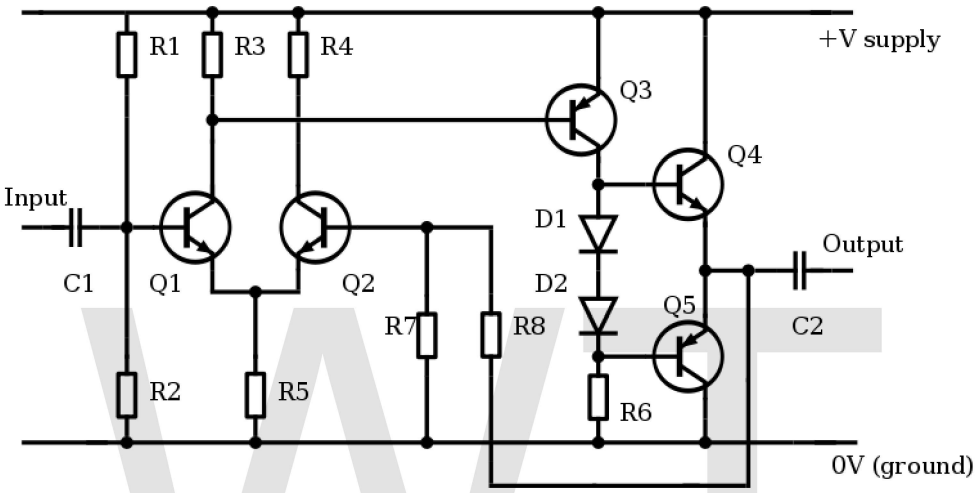
Several audio amplifier manufacturers have started "inventing" new classes as a way to differentiate themselves. These class names usually do not reflect any revolutionary amplification technique, and are used mostly for marketing purposes. This can easily be determined by the fact that the class name is trademarked or copyrighted. For example, Crown's K and I-Tech Series as well as several other models utilise Crown's patented Class-I (or BCA) technology. Lab.gruppen use a form of Class D amplifier called Class TD or Tracked Class D which tracks the waveform to more accurately amplify it without the drawbacks of traditional Class D amplifiers.

"Class T" was a trademark of TriPath company which manufactures audio amplifier ICs. This new Class T is a revision of the common Class D amplifier, but with changes to ensure fidelity over the full audio spectrum, unlike traditional Class D designs. It operates at different frequencies depending on the power output, with values ranging from as low as 200 kHz to 1.2 MHz, using a proprietary modulator. Tripath ceased operations in 2007, its patents acquired by Cirrus Logic for their Mixed-Signal Audio division.

"Class Z" is a trademark of Zetex Semiconductors (now part of Diodes Inc. of Dallas, TX) and is a direct-digital-feedback technology. Zetex-patented circuits are being utilised in the latest power amplifiers by NAD Electronics of Canada.

Amplifier circuit

The practical amplifier circuit below could be the basis for a moderate-power audio amplifier. It features a typical (though substantially simplified) design as found in modern amplifiers, with a class AB push–pull output stage, and uses some overall negative feedback. Bipolar transistors are shown, but this design would also be realizable with FETs or valves.



A practical amplifier circuit

The input signal is coupled through capacitor C1 to the base of transistor Q1. The capacitor allows the AC signal to pass, but blocks the DC bias voltage established by resistors R1 and R2 so that any preceding circuit is not affected by it. Q1 and Q2 form a differential amplifier (an amplifier that multiplies the difference between two inputs by some constant), in an arrangement known as a long-tailed pair. This arrangement is used to conveniently allow the use of negative feedback, which is fed from the output to Q2 via R7 and R8.

The negative feedback into the difference amplifier allows the amplifier to compare the input to the actual output. The amplified signal from Q1 is directly fed to the second stage, Q3, which is a common emitter stage that provides further amplification of the signal and the DC bias for the output stages, Q4 and Q5. R6 provides the load for Q3 (A better design would probably use some form of active load here, such as a constant-current sink). So far, all of the amplifier is operating in Class A. The output pair are arranged in Class AB push–pull, also called a complementary pair. They provide the majority of the current amplification (while consuming low quiescent current) and directly drive the load, connected via DC-blocking capacitor C2. The diodes D1 and D2 provide a small amount of constant voltage bias for the output pair, just biasing them into the conducting state so that crossover distortion is minimized. That is, the diodes push the output stage firmly into class-AB mode (assuming that the base-emitter drop of the output transistors is reduced by heat dissipation).

This design is simple, but a good basis for a practical design because it automatically stabilises its operating point, since feedback internally operates from DC up through the audio range and beyond. Further circuit elements would probably be found in a real design that would roll off the frequency response above the needed range to prevent the possibility of unwanted oscillation. Also, the use of fixed diode bias as shown here can cause problems if the diodes are not both electrically and thermally matched to the output transistors — if the output transistors turn on too much, they can easily overheat and destroy themselves, as the full current from the power supply is not limited at this stage.

A common solution to help stabilise the output devices is to include some emitter resistors, typically an ohm or so. Calculating the values of the circuit's resistors and capacitors is done based on the components employed and the intended use of the amp.

Notes on implementation

Real world amplifiers are imperfect.

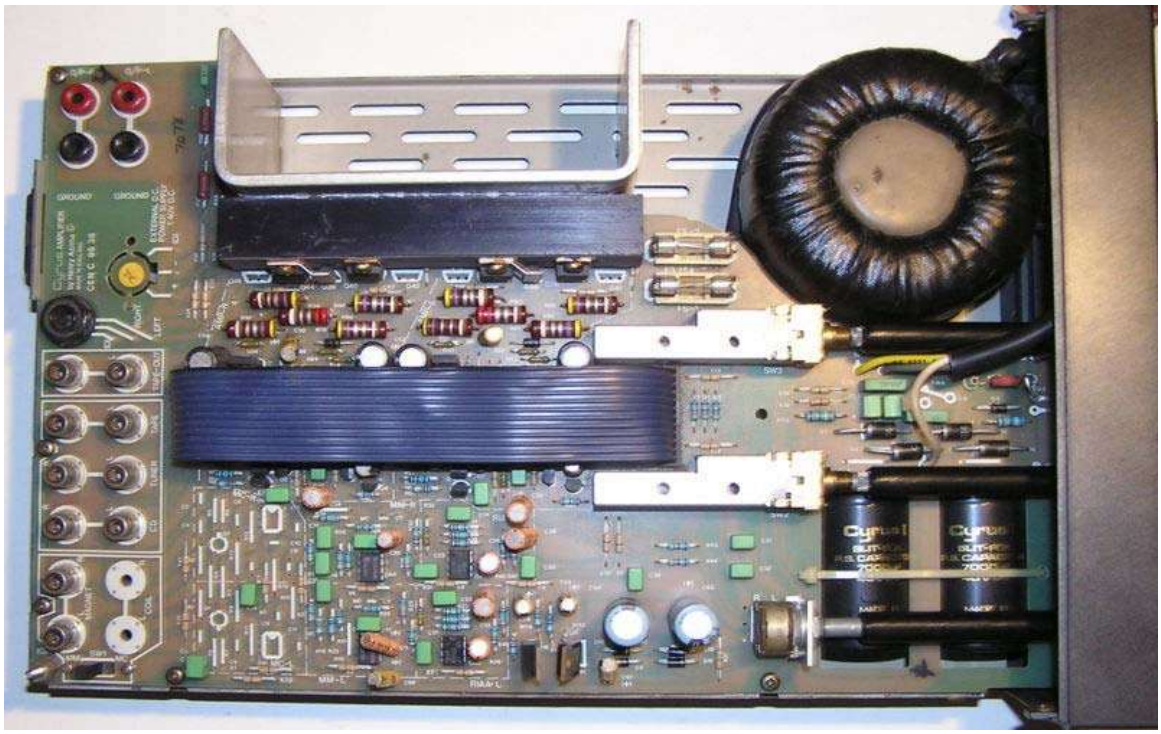
- One consequence is that the power supply itself may influence the output, and must itself be considered when designing the amplifier
- The amplifier circuit has an "open loop" performance, that can be described as various parameters (gain, slew rate, output impedance, distortion, bandwidth, signal to noise ratio, etc.)
- Many modern amplifiers use negative feedback techniques to hold the gain at the desired value.

Different methods of supplying power result in many different methods of bias. Bias is a technique by which the active devices are set up to operate in a particular regime, or by which the DC component of the output signal is set to the midpoint between the maximum voltages available from the power supply. Most amplifiers use several devices at each stage; they are typically matched in specifications except for polarity. Matched inverted polarity devices are called complementary pairs. Class A amplifiers generally use only one device, unless the power supply is set to provide both positive and negative voltages, in which case a dual device symmetrical design may be used. Class C amplifiers, by definition, use a single polarity supply.

Amplifiers often have multiple stages in cascade to increase gain. Each stage of these designs may be a different type of amp to suit the needs of that stage. For instance, the first stage might be a Class A stage, feeding a Class AB push–pull second stage, which then drives a Class G final output stage, taking advantage of the strengths of each type, while minimizing their weaknesses.

Chapter 8

Audio Amplifier



Mission Cyrus 1 Hi Fi integrated audio amplifier

An **audio amplifier** is an electronic amplifier that amplifies low-power audio signals (signals composed primarily of frequencies between 20 - 20 000 Hz, the human range of hearing) to a level suitable for driving loudspeakers and is the final stage in a typical audio playback chain.

The preceding stages in such a chain are low power audio amplifiers which perform tasks like pre-amplification, equalization, tone control, mixing/effects, or audio sources like record players, CD players, and cassette players. Most audio amplifiers require these low-level inputs to adhere to line levels.

While the input signal to an audio amplifier may measure only a few hundred microwatts, its output may be tens, hundreds, or thousands of watts.

History

Three audio amplifiers

The audio amplifier was invented in 1909 by Lee De Forest when he invented the triode vacuum tube. The triode was a three terminal device with a control grid that can modulate the flow of electrons from the filament to the plate. The triode vacuum amplifier was used to make the first AM radio.

Early audio amplifiers were based on vacuum tubes (also known as *valves*), and some of these achieved notably high quality (e.g., the Williamson amplifier of 1947-9). Most modern audio amplifiers are based on solid state devices (transistors such as BJTs, FETs and MOSFETs), but there are still some who prefer tube-based amplifiers, due to a perceived 'warmer' valve sound. Audio amplifiers based on transistors became practical with the wide availability of inexpensive transistors in the late 1960s.

Design parameters

Key design parameters for audio amplifiers are frequency response, gain, noise, and distortion. These are interdependent; increasing gain often leads to undesirable increases in noise and distortion. While negative feedback actually reduces the gain, it also reduces distortion. Most audio amplifiers are linear amplifiers operating in class AB.

Filters and preamplifiers

Historically, the majority of commercial audio preamplifiers made had complex filter circuits for equalization and tone adjustment, due to the far from ideal quality of recordings, playback technology, and speakers of the day.

Using today's high quality (often digital) source material, speakers, etc., such filter circuits are usually not needed. Audiophiles generally agree that filter circuits are to be avoided wherever possible. Today's audiophile amplifiers do not have tone controls or filters.

Since modern digital devices, including CD and DVD players, radio receivers and tape decks already provide a "flat" signal at line level, the preamp. is not needed other than as volume control. One alternative to a separate preamp is to simply use passive volume and switching controls, sometimes integrated into a power amp to form an "integrated" amplifier.

Further developments in amplifier design

For some years following the introduction of solid state amplifiers, their perceived sound did not have the excellent audio quality of the best valve amplifiers. This led audiophiles to believe that valve sound had an intrinsic quality due to the vacuum tube technology itself. In 1972, Matti Otala demonstrated the origin of a previously unobserved form of distortion: transitory intermodulation distortion (TIM), also called slew rate distortion. TIM distortion was found to occur during very rapid increases in amplifier output voltage. TIM did not appear at steady state sine tone measurements, helping to hide it from design engineers prior to 1972. Problems with TIM distortion stem from reduced open loop frequency response of solid state amplifiers. Further works of Otala and other authors found the solution for TIM distortion, including increasing slew rate, decreasing preamp frequency bandwidth, and the insertion of a lag compensation circuit in the input stage of the amplifier. In high quality modern amplifiers the open loop response is at least 20 kHz, canceling TIM distortion. However, TIM distortion is still present in most low price home quality amplifiers.

The next step in advanced design was the Baxandall Theorem, created by Peter Baxandall in England. This theorem introduced the concept of comparing the ratio between the input distortion and the output distortion of an audio amplifier. This new idea helped audio design engineers to better evaluate the distortion processes within an audio amplifier.

Applications

Important applications include public address systems, theatrical and concert sound reinforcement, and domestic sound systems. The sound card in a personal computer contains several audio amplifiers (depending on number of channels), as does every stereo or home-theatre system.

Chapter 9

Buffer Amplifier

A **buffer amplifier** (sometimes simply called a **buffer**) is one that provides electrical impedance transformation from one circuit to another. Two main types of buffer exist: the **voltage buffer** and the **current buffer**.

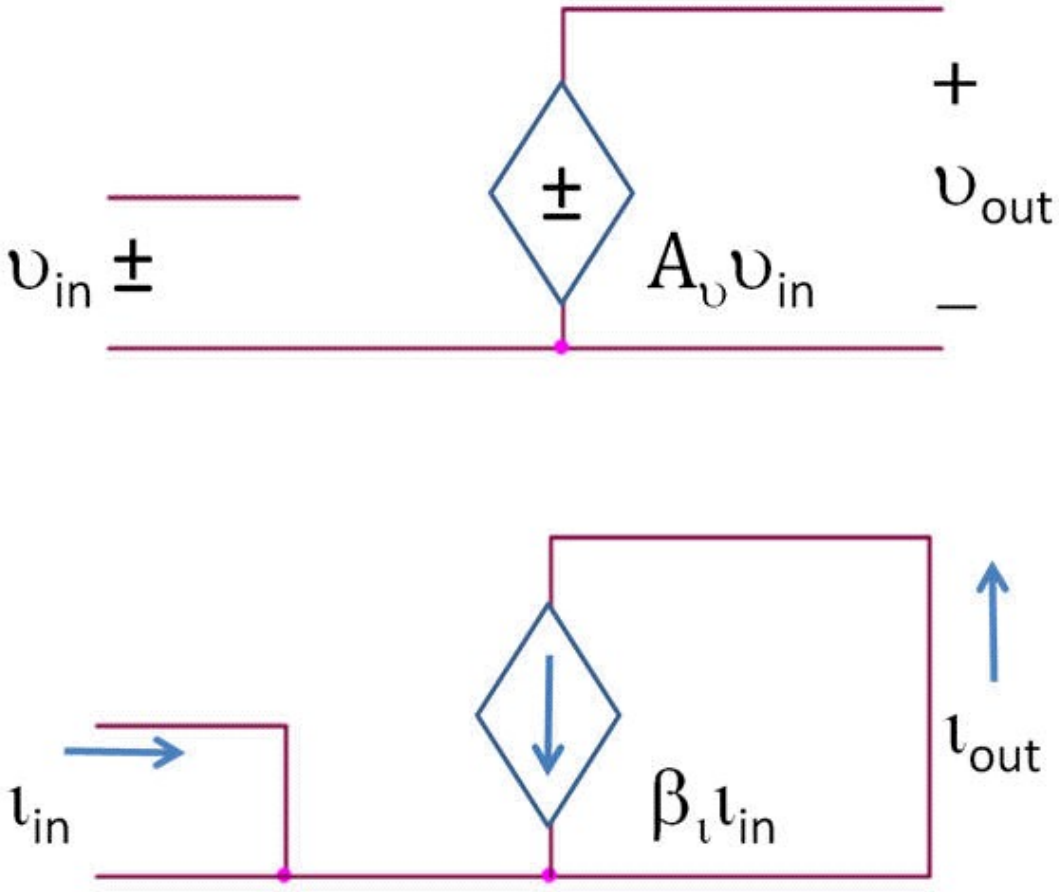


Figure 1: Top: Ideal voltage buffer Bottom: Ideal current buffer

Voltage buffer

A voltage buffer amplifier is used to transfer a voltage from a first circuit, having a high output impedance level, to a second circuit with a low input impedance level. The interposed buffer amplifier prevents the second circuit from loading the first circuit unacceptably and interfering with its desired operation. In the ideal voltage buffer in the diagram, the input resistance is infinite, the output resistance zero (impedance of an ideal voltage source is zero). Other properties of the ideal buffer are: perfect linearity, regardless of signal amplitudes; and instant output response, regardless of the speed of the input signal.

If the voltage is transferred unchanged (the voltage gain A_v is 1), the amplifier is a **unity gain buffer**; also known as a **voltage follower** because the output voltage *follows* or tracks the input voltage. Although the voltage gain of a voltage buffer amplifier may be (approximately) unity, it usually provides considerable current gain and thus power gain. However, it is commonplace to say that it has a gain of 1 (or the equivalent 0 dB), referring to the voltage gain.

As an example, consider a Thévenin source (voltage V_A, series resistance R_A) driving a resistor load R_L. Because of voltage division (also referred to as "loading") the voltage across the load is only $V_A R_L / (R_L + R_A)$. However, if the Thévenin source drives a unity gain buffer such as that in Figure 1 (top, with unity gain), the voltage input to the amplifier is V_A, with *no voltage division* because the amplifier input resistance is infinite. At the output the dependent voltage source delivers voltage $A_v V_A = V_A$ to the load, again without voltage division because the output resistance of the buffer is zero. A Thévenin equivalent circuit of the combined original Thévenin source *and* the buffer is an ideal voltage source V_A with zero Thévenin resistance.

Current buffer

Typically a current buffer amplifier is used to transfer a current from a first circuit, having a low output impedance level, to a second circuit with a high input impedance level. The interposed buffer amplifier prevents the second circuit from loading the first circuit unacceptably and interfering with its desired operation. In the ideal current buffer in the diagram, the input resistance is zero, the output resistance infinite (impedance of an ideal current source is infinite). Again, other properties of the ideal buffer are: perfect linearity, regardless of signal amplitudes; and instant output response, regardless of the speed of the input signal.

For a current buffer, if the current is transferred unchanged (the current gain β_i is 1), the amplifier is again a **unity gain buffer**; this time known as a **current follower** because the output current *follows* or tracks the input current.

As an example, consider a Norton source (current I_A, parallel resistance R_A) driving a resistor load R_L. Because of current division (also referred to as "loading") the current delivered to the load is only $I_A R_A / (R_L + R_A)$. However, if the Norton source drives a

unity gain buffer such as that in Figure 1 (bottom, with unity gain), the current input to the amplifier is I_A, with *no current division* because the amplifier input resistance is zero. At the output the dependent current source delivers current $\beta_i I_A = I_A$ to the load, again without current division because the output resistance of the buffer is infinite. A Norton equivalent circuit of the combined original Norton source *and* the buffer is an ideal current source I_A with infinite Norton resistance.

Voltage buffer examples

Op-amp implementation

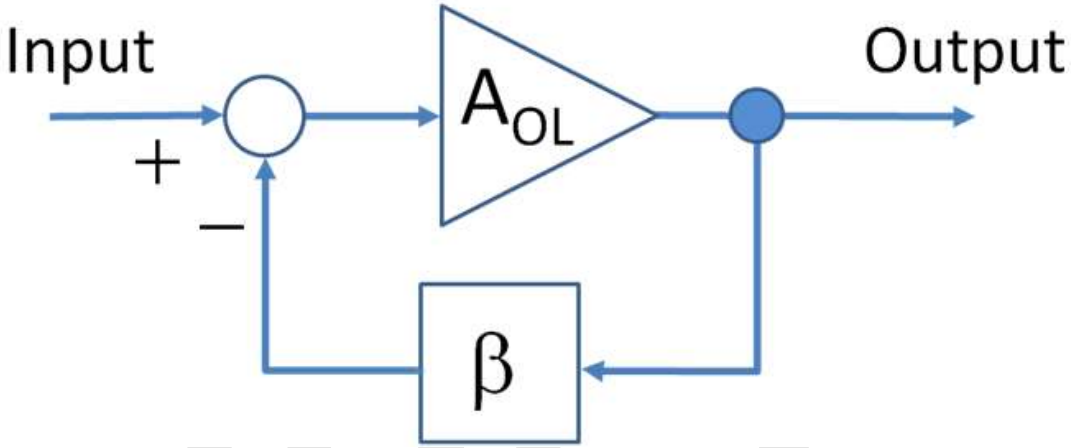


Figure 2: A negative feedback amplifier

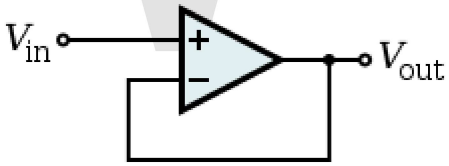


Figure 3. An op-amp–based unity gain buffer amplifier

A unity gain buffer amplifier may be constructed by applying a full series negative feedback (Fig. 2) to an op-amp simply by connecting its output to its inverting input, and connecting the signal source to the non-inverting input (Fig. 3). In this configuration, the entire output voltage (β = 1 in Fig. 2) is placed contrary and in series with the input voltage. Thus the two voltages are subtracted according to KVL and their difference is applied to the op-amp differential input. This connection forces the op-amp to adjust its output voltage simply equal to the input voltage (V_{out} follows V_{in} so the circuit is named op-amp voltage follower).

The importance of this circuit does not come from any change in voltage, but from the input and output impedances of the op-amp. The input impedance of the op-amp is very high (1 MΩ to 10 TΩ), meaning that the input of the op-amp does not load down the

source or draw any current from it. Because the output impedance of the op-amp is very low, it drives the load as if it were a perfect voltage source. Both the connections to and from the buffer are therefore bridging connections, which reduce power consumption in the source, distortion from overloading, crosstalk and other electromagnetic interference.

Single-transistor circuits

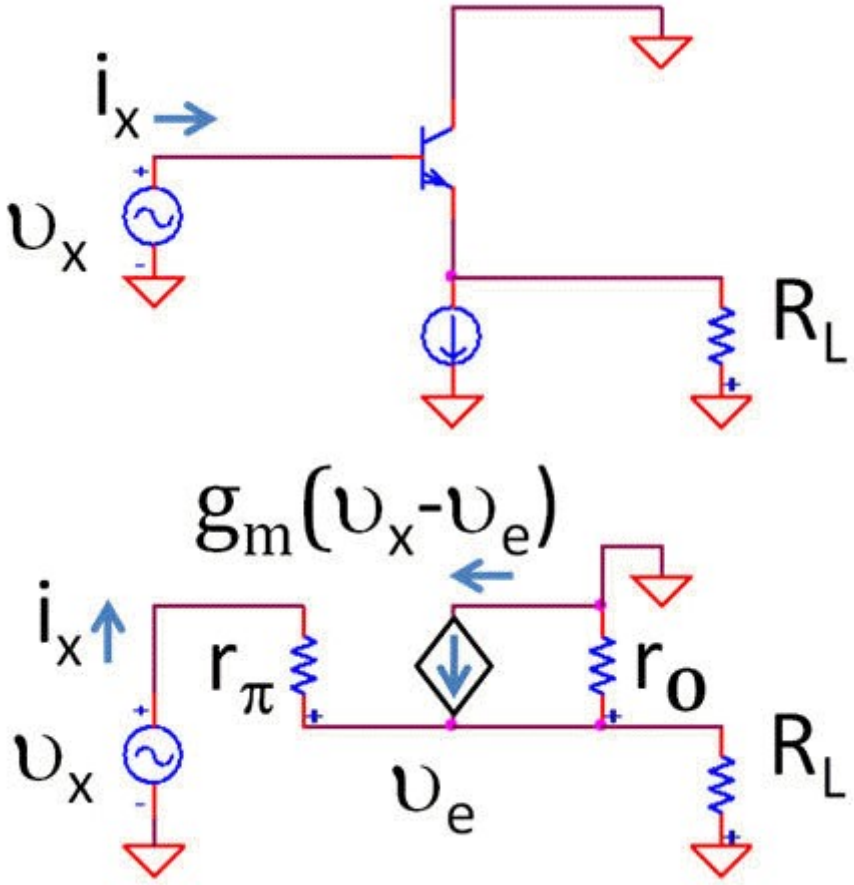


Figure 4: Top: BJT voltage follower Bottom: Small-signal, low-frequency equivalent circuit using hybrid-pi model

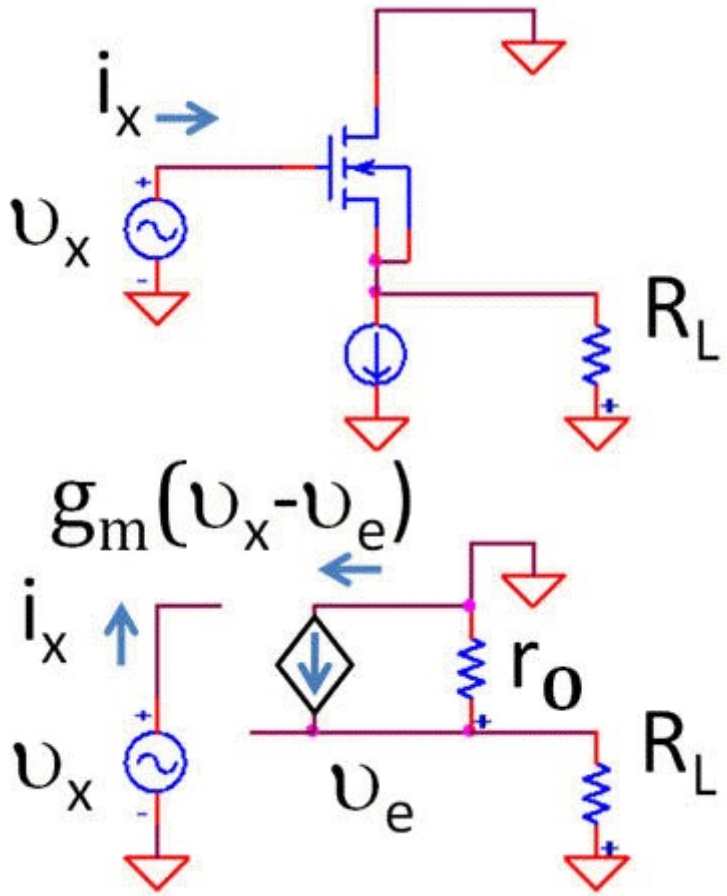


Figure 5: Top: MOSFET voltage follower Bottom: Small-signal, low-frequency equivalent circuit using hybrid-pi model

Other unity gain buffer amplifiers include the bipolar junction transistor in common-collector configuration (called an *emitter follower* because the emitter voltage follows the base voltage, or a *voltage follower* because the output voltage follows the input voltage); the field effect transistor in common-drain configuration (called a source follower because the source voltage follows the gate voltage or, again, a *voltage follower* because the output voltage follows the input voltage); or similar configurations using vacuum tubes (cathode follower), or other active devices. All such amplifiers actually have a gain of slightly less than unity, but the difference is usually small and unimportant.

Impedance transformation using the bipolar voltage follower

Using the small-signal circuit in Figure 4, the impedance seen looking into the circuit is

$$R_{in} = \frac{v_x}{i_x} = r_\pi + (\beta + 1)(r_O // R_L)$$

.

(The analysis uses the relation $g_m r_\pi = (I_C / V_T)\ (V_T / I_B) = \beta$, which follows from the evaluation of these parameters in terms of the bias currents.) Assuming the usual case where $r_O >> R_L$, the impedance is looking into the buffer is larger than the load R_L without the buffer by a factor of (β +1), which is substantial because β is large. The impedance is increased even more by the added r_π, but often $r_\pi << (\beta + 1)\ R_L$, so the addition does not make much difference.

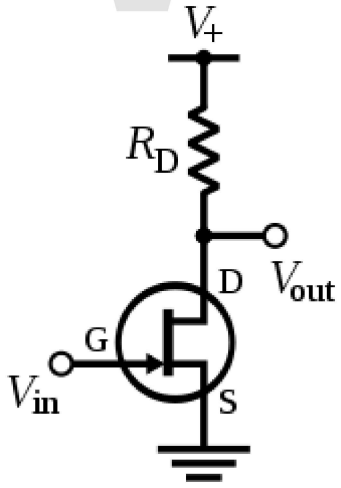
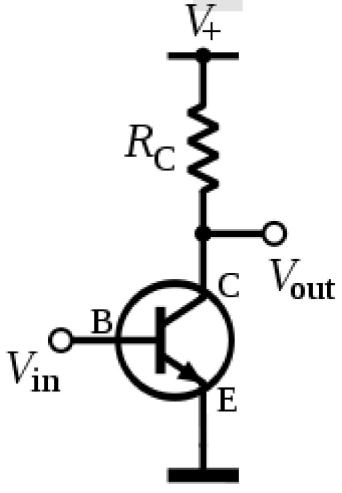
Impedance transformation using the MOSFET voltage follower

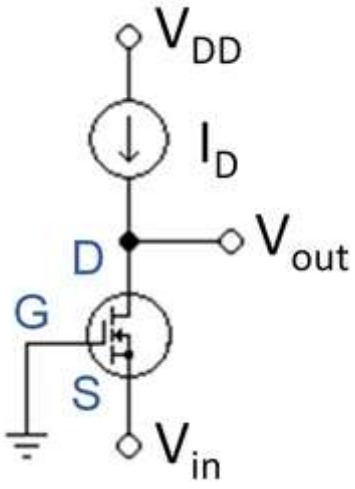
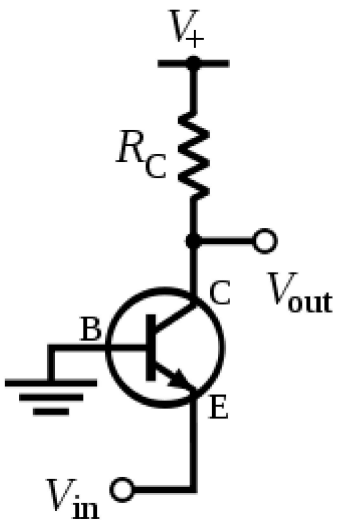
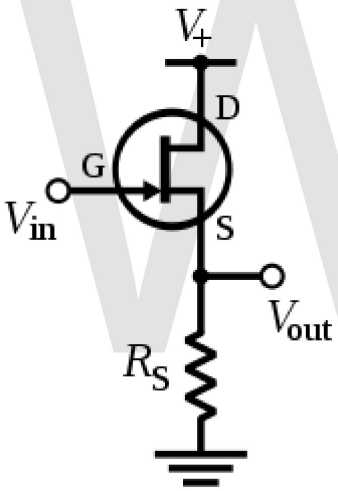
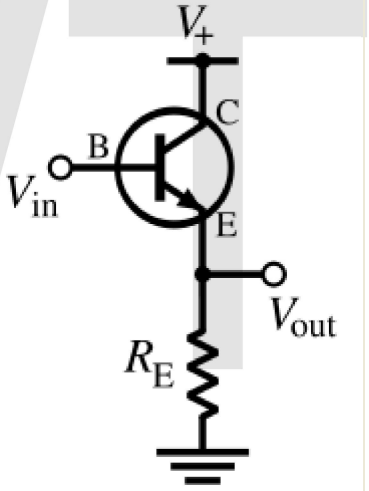
Using the small-signal circuit in Figure 5, the impedance seen looking into the circuit is no longer R_L but instead is infinite (at low frequencies) because the MOSFET draws no current.

As frequency is increased, the parasitic capacitances of the transistors come into play and the transformed input impedance drops with frequency.

Chart of Single-Transistor Amplifiers

All configurations of a single-transistor amplifier can be used as a buffer to isolate the driver from the load. For most digital applications, an NMOS voltage follower (common drain) is the preferred configuration; or an inverter (common source), if necessary. These amplifiers have high input impedance, which means that the digital system will not need to supply a large current.

Amplifier Type	MOSFET (NMOS)	BJT (npn)	Notes
Common Source / Common Emitter	 Common source	 Common emitter	Inverts the input signal

Common Gate / Common Base	 Common gate	 Common base	Typically used for current buffering (not voltage buffering); generally unsuitable for TTL voltage buffer
Common Drain / Common Collector	 Common drain	 Common collector	Unity-gain voltage buffer

Integrated buffer amplifiers

It is common for a single package to contain several discrete buffer amplifiers. For example, a **hex buffer** is a single package containing 6 discrete buffer amplifiers, and an **octal buffer** is a single package containing 8 discrete buffer amplifiers.

Speaker array amplifiers

The majority of amplifiers used to drive large speaker arrays, such as those used for rock concerts, are unity-gain, high-current amplifiers. Some current amplifiers take the voltage output from Class A/B, B, or tube (valve) amplifiers, while others contain built-in voltage amplifiers as a pre-amp stage. The result is a signal nearly identical to the input signal in

terms of voltage, but capable of sending high amounts of current into low impedance speaker arrays where the speakers are wired in parallel.

Current buffer examples

Simple unity gain buffer amplifiers include the bipolar junction transistor in common-base configuration, or the MOSFET in common-gate configuration (called a *current follower* because the output current follows the input current). The current gain of a current buffer amplifier is (approximately) unity.

Single-transistor circuits

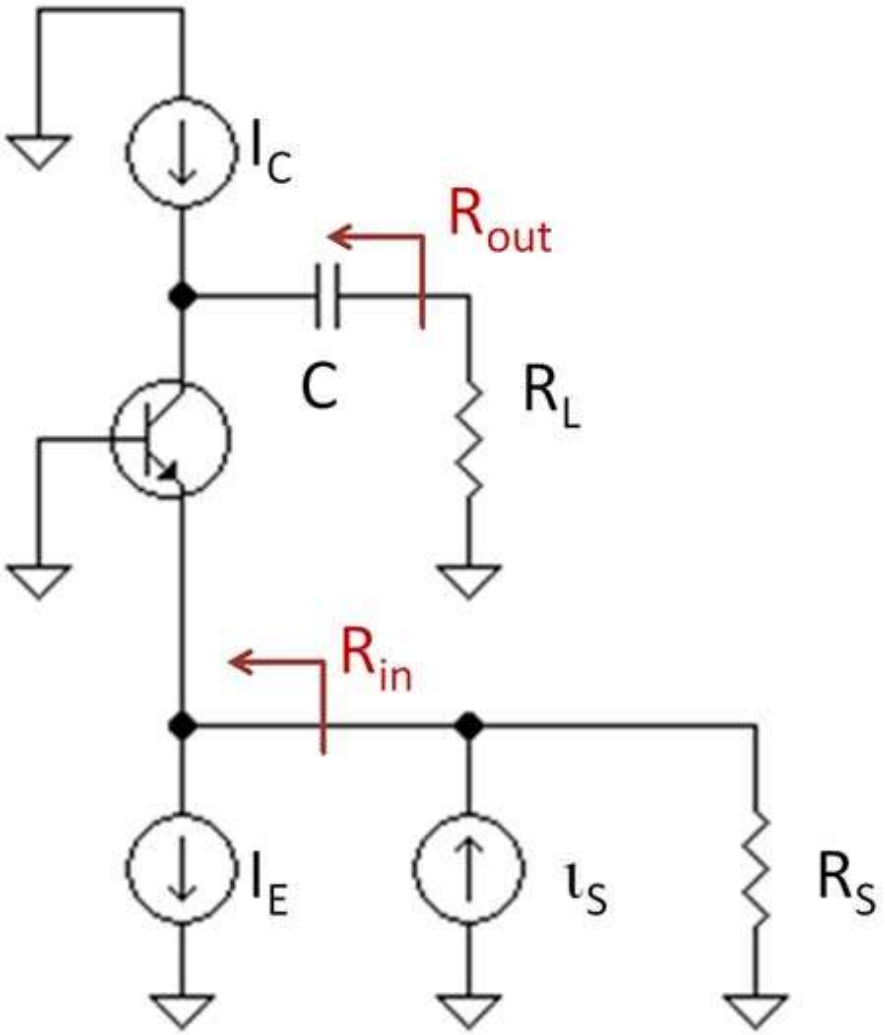


Figure 6: Bipolar current follower biased by current source I_E and with active load I_C

Figure 6 shows a bipolar current buffer biased with a current source (designated I_E for DC emitter current) and driving another DC current source as active load (designated I_C for DC collector current). The AC input signal current i_{in} is applied to the emitter node of the transistor by an AC Norton current source with Norton resistance R_S. The AC output current i_{out} is delivered by the buffer via a large coupling capacitor to load R_L. This coupling capacitor is large enough to be a short-circuit at frequencies of interest.

Because the transistor output resistance connects input and output sides of the circuit, there is a (very small) backward voltage feedback from the output to the input so this circuit is not unilateral. In addition, for the same reason, the input resistance depends (slightly) upon the output load resistance, and the output resistance depends significantly on the input driver resistance.

Chapter 10

Operational Amplifier

A Signetics μa741 operational amplifier, one of the most successful op-amps

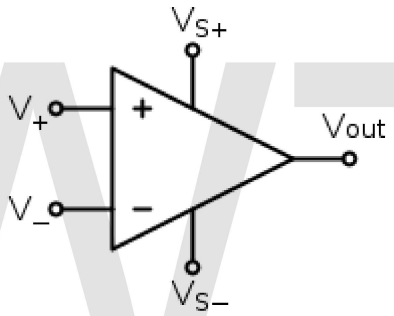
An **Operational amplifier** ("op-amp") is a DC-coupled high-gain electronic voltage amplifier with a differential input and, usually, a single-ended output. An op-amp produces an output voltage that is typically hundreds of thousands times larger than the voltage *difference* between its input terminals.

Operational amplifiers are important building blocks for a wide range of electronic circuits. They had their origins in analog computers where they were used in many linear, non-linear and frequency-dependent circuits. Their popularity in circuit design largely stems from the fact the characteristics of the final elements (such as their gain) are set by external components with little dependence on temperature changes and manufacturing variations in the op-amp itself.

Op-amps are among the most widely used electronic devices today, being used in a vast array of consumer, industrial, and scientific devices. Many standard IC op-amps cost only a few cents in moderate production volume; however some integrated or hybrid operational amplifiers with special performance specifications may cost over $100 US in small quantities. Op-amps may be packaged as components, or used as elements of more complex integrated circuits.

The op-amp is one type of differential amplifier. Other types of differential amplifier include the fully differential amplifier (similar to the op-amp, but with two outputs), the instrumentation amplifier (usually built from three op-amps), the isolation amplifier (similar to the instrumentation amplifier, but with tolerance to common-mode voltages that would destroy an ordinary op-amp), and negative feedback amplifier (usually built from one or more op-amps and a resistive feedback network).

Circuit notation



Circuit diagram symbol for an op-amp

The circuit symbol for an op-amp is shown to the right, where:

- V_+: non-inverting input
- V_-: inverting input
- $V_{\mathbf{out}}$: output
- V_{S+}: positive power supply
- V_{S-}: negative power supply

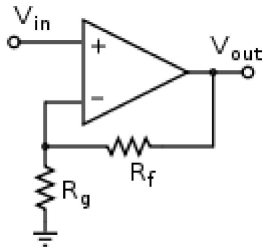
The power supply pins (V_{S+} and V_{S-}) can be labeled in different ways. Despite different labeling, the function remains the same — to provide additional power for amplification of the signal. Often these pins are left out of the diagram for clarity, and the power configuration is described or assumed from the circuit.

Operation

The amplifier's differential inputs consist of a V_+input and a V_-input, and ideally the op-amp amplifies only the difference in voltage between the two, which is called the *differential input voltage*. The output voltage of the op-amp is given by the equation,

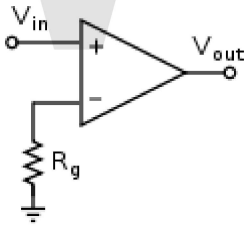
$$V_{\text{out}} = (V_+ - V_-)\, A_{OL}$$

where V_+ is the voltage at the non-inverting terminal, V_- is the voltage at the inverting terminal and A_{OL} is the open-loop gain of the amplifier. (The term "open-loop" refers to the absence of a feedback loop from the output to the input.)



Typically the op-amp's very large gain is controlled by negative feedback, which largely determines the magnitude of its output ("closed-loop") voltage gain in amplifier applications, or the transfer function required (in analog computers). Without negative feedback, and perhaps with positive feedback for regeneration, an op-amp acts as a comparator. High input impedance at the input terminals and low output impedance at the output terminal(s) are important typical characteristics.

With no negative feedback, the op-amp acts as a comparator. The inverting input is held at ground (0 V) by the resistor, so if the V_{in} applied to the non-inverting input is positive, the output will be maximum positive, and if V_{in} is negative, the output will be maximum negative. Since there is no feedback from the output to either input, this is an *open loop* circuit. The circuit's gain is just the G_{OL} of the op-amp.



Adding negative feedback via the voltage divider R_f,R_g reduces the gain. Equilibrium will be established when V_{out} is just sufficient to reach around and "pull" the inverting input to the same voltage as V_{in}. As a simple example, if V_{in} = 1 V and $R_f = R_g$, V_{out} will be 2 V, the amount required to keep V_- at 1 V. Because of the feedback provided by R_f,R_g this is a *closed loop* circuit. Its over-all gain V_{out} / V_{in} is called the *closed-loop gain* A_{CL}. Because the feedback is negative, in this case A_{CL} is less than the A_{OL} of the op-amp.

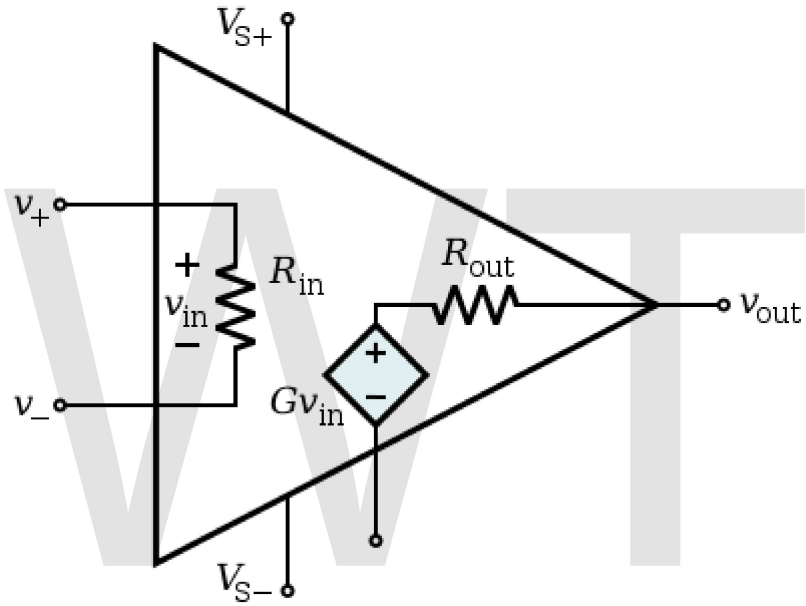
The magnitude of A_{OL} is typically very large—10,000 or more for integrated circuit op-amps—and therefore even a quite small difference between V_+ and V_- drives the amplifier output nearly to the supply voltage. This is called *saturation* of the amplifier. The magnitude of A_{OL} is not well controlled by the manufacturing process, and so it is impractical to use an operational amplifier as a stand-alone differential amplifier. If predictable operation is desired, negative feedback is used, by applying a portion of the

output voltage to the inverting input. The *closed loop* feedback greatly reduces the gain of the amplifier. If negative feedback is used, the circuit's overall gain and other parameters become determined more by the feedback network than by the op-amp itself. If the feedback network is made of components with relatively constant, stable values, the unpredictability and inconstancy of the op-amp's parameters do not seriously affect the circuit's performance.

If no negative feedback is used, the op-amp functions as a switch or comparator.

Positive feedback may be used to introduce hysteresis or oscillation.

Ideal and real op-amps



An equivalent circuit of an operational amplifier that models some resistive non-ideal parameters.

An ideal op-amp is usually considered to have the following properties, and they are considered to hold for all input voltages:

- Infinite open-loop gain (when doing theoretical analysis, a limit may be taken as open loop gain A_{OL} goes to infinity).
- Infinite voltage range available at the output (v_{out}) (in practice the voltages available from the output are limited by the supply voltages V_{S+} and V_{S-}). The power supply sources are called rails.
- Infinite bandwidth (i.e., the frequency magnitude response is considered to be flat everywhere with zero phase shift).
- Infinite input impedance (so, in the diagram, $R_{in} = \infty$, and zero current flows from v_+ to v_-).

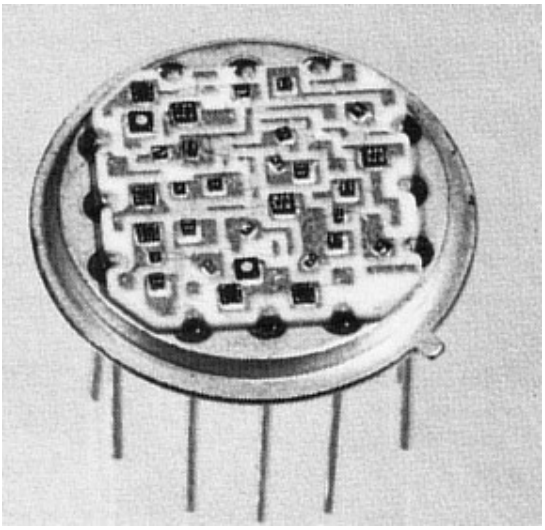
- Zero input current (i.e., there is assumed to be no leakage or bias current into the device).
- Zero input offset voltage (i.e., when the input terminals are shorted so that $v_+ = v_-$, the output is a virtual ground or $v_{out} = 0$).
- Infinite slew rate (i.e., the rate of change of the output voltage is unbounded) and power bandwidth (full output voltage and current available at all frequencies).
- Zero output impedance (i.e., $R_{out} = 0$, so that output voltage does not vary with output current).
- Zero noise.
- Infinite Common-mode rejection ratio (CMRR).
- Infinite Power supply rejection ratio for both power supply rails.

In practice, none of these ideals can be realized, and various shortcomings and compromises have to be accepted. Depending on the parameters of interest, a real op-amp may be modeled to take account of some of the non-infinite or non-zero parameters using equivalent resistors and capacitors in the op-amp model. The designer can then include the effects of these undesirable, but real, effects into the overall performance of the final circuit. Some parameters may turn out to have negligible effect on the final design while others represent actual limitations of the final performance, that must be evaluated.

History

An op-amp in a modern DIP

GAP/R's K2-W: a vacuum-tube op-amp (1953)



ADI's HOS-050: a high speed hybrid IC op-amp (1979)

1941: First (vacuum tube) op-amp

An op-amp, defined as a general-purpose, DC-coupled, high gain, inverting feedback amplifier, is first found in U.S. Patent 2,401,779 "Summing Amplifier" filed by Karl D. Swartzel Jr. of Bell labs in 1941. This design used three vacuum tubes to achieve a gain of 90 dB and operated on voltage rails of ±350 V. It had a single inverting input rather than differential inverting and non-inverting inputs, as are common in today's op-amps. Throughout World War II, Swartzel's design proved its value by being liberally used in the M9 artillery director designed at Bell Labs. This artillery director worked with the SCR584 radar system to achieve extraordinary hit rates (near 90%) that would not have been possible otherwise.

1947: First op-amp with an explicit non-inverting input

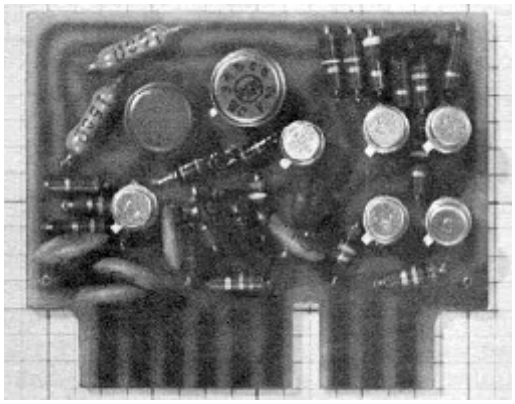
In 1947, the operational amplifier was first formally defined and named in a paper by Professor John R. Ragazzini of Columbia University. In this same paper a footnote mentioned an op-amp design by a student that would turn out to be quite significant. This op-amp, designed by Loebe Julie, was superior in a variety of ways. It had two major innovations. Its input stage used a long-tailed triode pair with loads matched to reduce drift in the output and, far more importantly, it was the first op-amp design to have two inputs (one inverting, the other non-inverting). The differential input made a whole range of new functionality possible, but it would not be used for a long time due to the rise of the chopper-stabilized amplifier.

1949: First chopper-stabilized op-amp

In 1949, Edwin A. Goldberg designed a chopper-stabilized op-amp. This set-up uses a normal op-amp with an additional AC amplifier that goes alongside the op-amp. The chopper gets an AC signal from DC by switching between the DC voltage and ground at a fast rate (60 Hz or 400 Hz). This signal is then amplified, rectified, filtered and fed into the op-amp's non-inverting input. This vastly improved the gain of the op-amp while significantly reducing the output drift and DC offset. Unfortunately, any design that used a chopper couldn't use their non-inverting input for any other purpose. Nevertheless, the much improved characteristics of the chopper-stabilized op-amp made it the dominant way to use op-amps. Techniques that used the non-inverting input regularly would not be very popular until the 1960s when op-amp ICs started to show up in the field.

In 1953, vacuum tube op-amps became commercially available with the release of the model K2-W from George A. Philbrick Researches, Incorporated. The designation on the devices shown, GAP/R, is a contraction for the complete company name. Two nine-pin 12AX7 vacuum tubes were mounted in an octal package and had a model K2-P chopper add-on available that would effectively "use up" the non-inverting input. This op-amp was based on a descendant of Loebe Julie's 1947 design and, along with its successors, would start the widespread use of op-amps in industry.

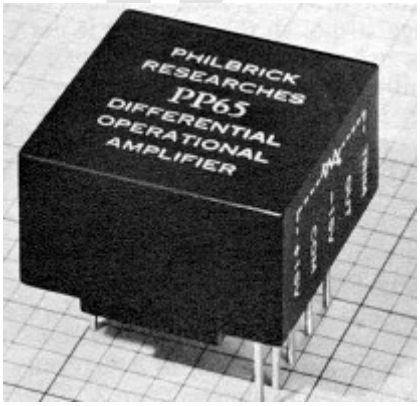
1961: First discrete IC op-amps



GAP/R's model P45: a solid-state, discrete op-amp (1961)

With the birth of the transistor in 1947, and the silicon transistor in 1954, the concept of ICs became a reality. The introduction of the planar process in 1959 made transistors and ICs stable enough to be commercially useful. By 1961, solid-state, discrete op-amps were being produced. These op-amps were effectively small circuit boards with packages such as edge-connectors. They usually had hand-selected resistors in order to improve things such as voltage offset and drift. The P45 (1961) had a gain of 94 dB and ran on ±15 V rails. It was intended to deal with signals in the range of ±10 V.

1962: First op-amps in potted modules



GAP/R's model PP65: a solid-state op-amp in a potted module (1962)

By 1962, several companies were producing modular potted packages that could be plugged into printed circuit boards. These packages were crucially important as they made the operational amplifier into a single black box which could be easily treated as a component in a larger circuit.

1963: First monolithic IC op-amp

In 1963, the first monolithic IC op-amp, the µA702 designed by Bob Widlar at Fairchild Semiconductor, was released. Monolithic ICs consist of a single chip as opposed to a chip and discrete parts (a discrete IC) or multiple chips bonded and connected on a circuit board (a hybrid IC). Almost all modern op-amps are monolithic ICs; however, this first IC did not meet with much success. Issues such as an uneven supply voltage, low gain and a small dynamic range held off the dominance of monolithic op-amps until 1965 when the µA709 (also designed by Bob Widlar) was released.

1966: First varactor bridge op-amps

Since the 741, there have been many different directions taken in op-amp design. Varactor bridge op-amps started to be produced in the late 1960s. They were designed to have extremely small input current and are still amongst the best op-amps available in terms of common-mode rejection with the ability to correctly deal with hundreds of volts at their inputs.

1968: Release of the µA741

The popularity of monolithic op-amps was further improved upon the release of the LM101 in 1967, which solved a variety of issues, and the subsequent release of the µA741 in 1968. The µA741 was extremely similar to the LM101 except that Fairchild's facilities allowed them to include a 30 pF compensation capacitor inside the chip instead of requiring external compensation. This simple difference has made the 741 *the* canonical op-amp and many modern amps base their pinout on the 741s.The µA741 is still in production, and has become ubiquitous in electronics—many manufacturers produce a version of this classic chip, recognizable by part numbers containing *741*.

1970: First high-speed, low-input current FET design

In the 1970s high speed, low-input current designs started to be made by using FETs. These would be largely replaced by op-amps made with MOSFETs in the 1980s. During the 1970s single sided supply op-amps also became available.

1972: Single sided supply op-amps being produced

A single sided supply op-amp is one where the input and output voltages can be as low as the negative power supply voltage instead of needing to be at least two volts above it. The result is that it can operate in many applications with the negative supply pin on the op-amp being connected to the signal ground, thus eliminating the need for a separate negative power supply.

The LM324 (released in 1972) was one such op-amp that came in a quad package (four separate op-amps in one package) and became an industry standard. In addition to packaging multiple op-amps in a single package, the 1970s also saw the birth of op-amps

in hybrid packages. These op-amps were generally improved versions of existing monolithic op-amps. As the properties of monolithic op-amps improved, the more complex hybrid ICs were quickly relegated to systems that are required to have extremely long service lives or other specialty systems.

Recent trends

Recently supply voltages in analog circuits have decreased (as they have in digital logic) and low-voltage opamps have been introduced reflecting this. Supplies of ±5V and increasingly 5V are common. To maximize the signal range modern op-amps commonly have rail-to-rail outputs and sometimes rail-to-rail inputs (the input signals can range from the lowest supply voltage to the highest).

Classification

Op-amps may be classified by their construction:

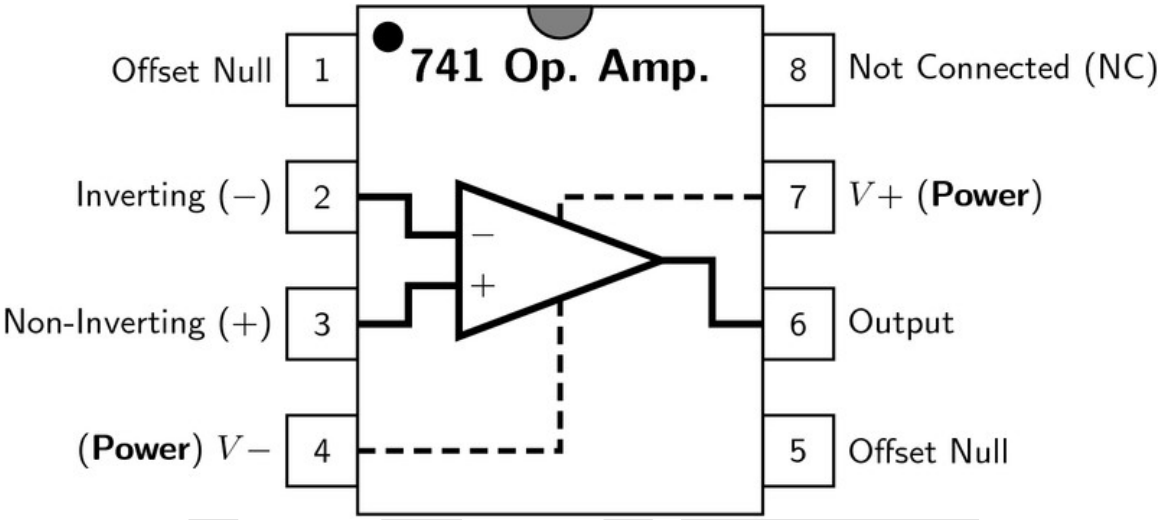
- discrete (built from individual transistors or tubes/valves)
- IC (fabricated in an Integrated circuit) - most common
- hybrid

IC op-amps may be classified in many ways, including:

- Military, Industrial, or Commercial grade (for example: the LM301 is the commercial grade version of the LM101, the LM201 is the industrial version). This may define operating temperature ranges and other environmental or quality factors.
- Classification by package type may also affect environmental hardiness, as well as manufacturing options; DIP, and other through-hole packages are tending to be replaced by Surface-mount devices.
- Classification by internal compensation: op-amps may suffer from high frequency instability in some negative feedback circuits unless a small compensation capacitor modifies the phase- and frequency- responses; op-amps with capacitor built in are termed "*compensated*", or perhaps compensated for closed-loop gains down to (say) 5, others: uncompensated.
- Single, dual and quad versions of many commercial op-amp IC are available, meaning 1, 2 or 4 operational amplifiers are included in the same package.
- Rail-to-rail input (and/or output) op-amps can work with input (and/or output) signals very close to the power supply rails.
- CMOS op-amps (such as the CA3140E) provide extremely high input resistances, higher than JFET-input op-amps, which are normally higher than bipolar-input op-amps.
- other varieties of op-amp include programmable op-amps (simply meaning the quiescent current, gain, bandwidth and so on can be adjusted slightly by an external resistor).

- manufacturers often tabulate their op-amps according to purpose, such as low-noise pre-amplifiers, wide bandwidth amplifiers, and so on.

Applications



DIP pinout for 741-type operational amplifier

Use in electronics system design

The use of op-amps as circuit blocks is much easier and clearer than specifying all their individual circuit elements (transistors, resistors, etc.), whether the amplifiers used are integrated or discrete. In the first approximation op-amps can be used as if they were ideal differential gain blocks; at a later stage limits can be placed on the acceptable range of parameters for each op-amp.

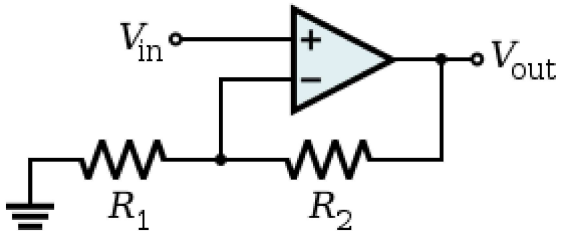
Circuit design follows the same lines for all electronic circuits. A specification is drawn up governing what the circuit is required to do, with allowable limits. For example, the gain may be required to be 100 times, with a tolerance of 5% but drift of less than 1% in a specified temperature range; the input impedance not less than one megohm; etc.

A basic circuit is designed, often with the help of circuit modeling (on a computer). Specific commercially available op-amps and other components are then chosen that meet the design criteria within the specified tolerances at acceptable cost. If not all criteria can be met, the specification may need to be modified.

A prototype is then built and tested; changes to meet or improve the specification, alter functionality, or reduce the cost, may be made.

Basic single stage amplifiers

Non-inverting amplifier



An op-amp connected in the non-inverting amplifier configuration

In a non-inverting amplifier, the output voltage changes in the same direction as the input voltage.

The gain equation for the op-amp is:

$$V_{out} = (V_+ - V_-)\, A_{OL}$$

However, in this circuit V_- is a function of V_{out} because of the negative feedback through the R_1R_2 network. R_1 and R_2 form a voltage divider, and as V_- is a high-impedance input, it does not load it appreciably. Consequently:

$$V_- = \beta \cdot V_{out}$$

where

$$\beta = \frac{R_1}{R_1 + R_2}$$

Substituting this into the gain equation, we obtain:

$$V_{out} = (V_{in} - \beta \cdot V_{out}) \cdot A_{OL}$$

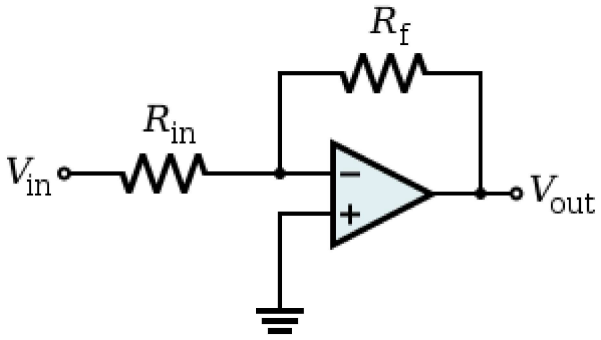
Solving for V_{out}:

$$V_{out} = V_{in} \cdot (\frac{1}{\beta + 1/A_{OL}})$$

If A_{OL} is very large, this simplifies to

$$V_{\text{out}} \approx \frac{V_{\text{in}}}{\beta} = \frac{V_{\text{in}}}{\frac{R_1}{R_1+R_2}} = V_{\text{in}}(1+\frac{R_2}{R_1})$$.

Inverting amplifier



An op-amp connected in the inverting amplifier configuration

In an inverting amplifier, the output voltage changes in an opposite direction to the input voltage.

As for the non-inverting amplifier, we start with the gain equation of the op-amp:

$$V_{\text{out}} = (V_+ - V_-)\, A_{OL}$$

This time, V_- is a function of both V_{out} and V_{in} due to the voltage divider formed by R_{f} and R_{in}. Again, the op-amp input does not apply an appreciable load, so:

$$V_- = \frac{1}{R_{\text{f}} + R_{\text{in}}}\left(R_{\text{f}} V_{\text{in}} + R_{\text{in}} V_{\text{out}}\right)$$

Substituting this into the gain equation and solving for V_{out}:

$$V_{\text{out}} = -V_{\text{in}} \cdot \frac{A_{OL} R_{\text{f}}}{R_{\text{f}} + R_{\text{in}} + A_{OL} R_{\text{in}}}$$

If A_{OL} is very large, this simplifies to

$$V_{\text{out}} \approx -V_{\text{in}} \frac{R_{\text{f}}}{R_{\text{in}}}.$$

A resistor is often inserted between the non-inverting input and ground (so both inputs "see" similar resistances), reducing the input offset voltage due to different voltage drops due to bias current, and may reduce distortion in some op-amps.

A DC-blocking capacitor may be inserted in series with the input resistor when a frequency response down to DC is not needed and any DC voltage on the input is unwanted. That is, the capacitive component of the input impedance inserts a DC zero and a low-frequency pole that gives the circuit a bandpass or high-pass characteristic.

Positive feedback configurations

Another typical configuration of op-amps is the positive feedback, which takes a fraction of the output signal back to the non-inverting input. An important application of it is the comparator with hysteresis (i.e., the Schmitt trigger).

Positive Voltage Level Detector

A positive reference voltage V_{ref} is applied to one of the op amp's inputs. This means that the op amp is set up as a comparator to detect a positive voltage. If the voltage to be sensed. E_i, is applied to op amp's (+) input, the result is a noninverting positive-level detector. When E_i is above V_{ref}, V_O equals + V_{sat}. When E_i is below V_{ref}, V_O equals -V_{sat}.

If E_i, is applied to the inverting input, the circuit is an inverting positive-level detector. Its operation can be summarized by the statement: When E_i is above V_{ref}, V_O equals -V_{sat}. This Circuit action can be seen more clearly by observing the plot E_i and V_{ref} versus time.

Negative Voltage Level Detector

A negative voltage detector is a circuit that detects when input signal E_i crosses the negative voltage -V_{ref}. When E_i is above -V_{ref}, V_O equals +V_{sat}. When $E_i$is below -V_{ref}, V_O equals -V_{sat}.When E_i is above -V_{ref}, V_O equals -V_{sat}, and when $E_i$is below -V_{ref}, V_O equals +V_{sat}.

Sine to Square Wave Converter

The zero detector will convert the output of a sine-wave from a function generator into a variable-frequency square wave. If E_i is a sine wave, triangular wave, or wave of any other shape that is symmetrical around zero, the zero- crossing detector's output will be square. The frequency of E_i should be below 100Hz.

Other applications

- audio- and video-frequency pre-amplifiers and buffers
- voltage comparators
- differential amplifiers
- differentiators and integrators
- filters
- precision rectifiers
- precision peak detectors
- voltage and current regulators

- analog calculators
- analog-to-digital converters
- digital-to-analog converter
- voltage clamps
- oscillators and waveform generators

Most single, dual and quad op-amps available have a standardized pin-out which permits one type to be substituted for another without wiring changes. A specific op-amp may be chosen for its open loop gain, bandwidth, noise performance, input impedance, power consumption, or a compromise between any of these factors.

Limitations of real op-amps

Real op-amps differ from the ideal model in various respects.

IC op-amps as implemented in practice are moderately complex integrated circuits; the internal circuitry for the relatively simple 741 op-amp below, for example.

DC imperfections

Real operational amplifiers suffer from several non-ideal effects:

Finite gain
: Open-loop gain is infinite in the ideal operational amplifier but finite in real operational amplifiers. Typical devices exhibit open-loop DC gain ranging from 100,000 to over 1 million. So long as the loop gain (i.e., the product of open-loop and feedback gains) is very large, the circuit gain will be determined entirely by the amount of negative feedback (i.e., it will be independent of open-loop gain). In cases where closed-loop gain must be very high, the feedback gain will be very low, and the low feedback gain causes low loop gain; in these cases, the operational amplifier will cease to behave ideally.

Finite input impedances
: The *differential input impedance* of the operational amplifier is defined as the impedance *between* its two inputs; the *common-mode input impedance* is the impedance from each input to ground. MOSFET-input operational amplifiers often have protection circuits that effectively short circuit any input differences greater than a small threshold, so the input impedance can appear to be very low in some tests. However, as long as these operational amplifiers are used in a typical high-gain negative feedback application, these protection circuits will be inactive. The input bias and leakage currents described below are a more important design parameter for typical operational amplifier applications.

Non-zero output impedance
: Low output impedance is important for low-impedance loads; for these loads, the voltage drop across the output impedance of the amplifier will be significant. Hence, the output impedance of the amplifier limits the maximum power that can be provided. In a negative-feedback configuration, the output impedance of the

amplifier is effectively lowered; thus, in linear applications, op-amps usually exhibit a very low output impedance indeed. Negative feedback can not, however, reduce the limitations that R_{load} in conjunction with R_{out} place on the maximum and minimum possible output voltages; it can only reduce output errors *within* that range.

Low-impedance outputs typically require high quiescent (i.e., idle) current in the output stage and will dissipate more power, so low-power designs may purposely sacrifice low output impedance.

Input current

Due to biasing requirements or leakage, a small amount of current (typically ~10 nanoamperes for bipolar op-amps, tens of picoamperes for JFET input stages, and only a few pA for MOSFET input stages) flows into the inputs. When large resistors or sources with high output impedances are used in the circuit, these small currents can produce large unmodeled voltage drops. If the input currents are matched, *and* the impedance looking *out* of *both* inputs are matched, then the voltages produced at each input will be equal. Because the operational amplifier operates on the *difference* between its inputs, these matched voltages will have no effect (unless the operational amplifier has poor CMRR, which is described below). It is more common for the input currents (or the impedances looking out of each input) to be slightly mismatched, and so a small *offset voltage* can be produced. This offset voltage can create offsets or drifting in the operational amplifier. It can often be nulled externally; however, many operational amplifiers include *offset null* or *balance* pins and some procedure for using them to remove this offset. Some operational amplifiers attempt to nullify this offset automatically.

Input offset voltage

This voltage, which is what is required across the op-amp's input terminals to drive the output voltage to zero, is related to the mismatches in input bias current. In the perfect amplifier, there would be no input offset voltage. However, it exists in actual op-amps because of imperfections in the differential amplifier that constitutes the input stage of the vast majority of these devices. Input offset voltage creates two problems: First, due to the amplifier's high voltage gain, it virtually assures that the amplifier output will go into saturation if it is operated without negative feedback, even when the input terminals are wired together. Second, in a closed loop, negative feedback configuration, the input offset voltage is amplified along with the signal and this may pose a problem if high precision DC amplification is required or if the input signal is very small.

Common mode gain

A perfect operational amplifier amplifies only the voltage difference between its two inputs, completely rejecting all voltages that are common to both. However, the differential input stage of an operational amplifier is never perfect, leading to the amplification of these identical voltages to some degree. The standard measure of this defect is called the common-mode rejection ratio (denoted CMRR). Minimization of common mode gain is usually important in non-inverting amplifiers (described below) that operate at high amplification.

Temperature effects
: All parameters change with temperature. Temperature drift of the input offset voltage is especially important.

Power-supply rejection
: The output of a perfect operational amplifier will be completely independent from ripples that arrive on its power supply inputs. Every real operational amplifier has a specified power supply rejection ratio (PSRR) that reflects how well the op-amp can reject changes in its supply voltage. Copious use of bypass capacitors can improve the PSRR of many devices, including the operational amplifier.

Drift
: Real op-amp parameters are subject to slow change over time and with changes in temperature, input conditions, etc.

Noise
: Amplifiers generate random voltage at the output even when there is no signal applied. This can be due to thermal noise and flicker noise of the devices. For applications with high gain or high bandwidth, noise becomes a very important consideration.

AC imperfections

The op-amp gain calculated at DC does not apply at higher frequencies. To a first approximation, the gain of a typical op-amp is inversely proportional to frequency. This means that an op-amp is characterized by its gain-bandwidth product. For example, an op-amp with a gain bandwidth product of 1 MHz would have a gain of 5 at 200 kHz, and a gain of 1 at 1 MHz. This low-pass characteristic is introduced deliberately, because it tends to stabilize the circuit by introducing a dominant pole. This is known as frequency compensation.

Typical low cost, general purpose op-amps exhibit a gain bandwidth product of a few megahertz. Specialty and high speed op-amps can achieve gain bandwidth products of hundreds of megahertz. For very high-frequency circuits, a completely different form of op-amp called the current-feedback operational amplifier is often used.

Other imperfections include:

Finite bandwidth
: All amplifiers have a finite bandwidth. This creates several problems for op amps. First, associated with the bandwidth limitation is a phase difference between the input signal and the amplifier output that can lead to oscillation in some feedback circuits. The internal frequency compensation used in some op amps to increase the gain or phase margin intentionally reduces the bandwidth even further to maintain output stability when using a wide variety of feedback networks. Second, reduced bandwidth results in lower amounts of feedback at higher frequencies, producing higher distortion, noise, and output impedance and also reduced output phase linearity as the frequency increases.

Input capacitance

Most important for high frequency operation because it further reduces the open loop bandwidth of the amplifier.

Common mode gain

Non-linear imperfections

Saturation

output voltage is limited to a minimum and maximum value close to the power supply voltages. Saturation occurs when the output of the amplifier reaches this value and is usually due to:

- In the case of an op-amp using a bipolar power supply, a voltage gain that produces an output that is more positive or more negative than that maximum or minimum; or
- In the case of an op-amp using a single supply voltage, either a voltage gain that produces an output that is more positive than that maximum, or a signal so close to ground that the amplifier's gain is not sufficient to raise it above the lower threshold.

Slewing

the amplifier's output voltage reaches its maximum rate of change. Measured as the slew rate, it is usually specified in volts per microsecond. When slewing occurs, further increases in the input signal have no effect on the rate of change of the output. Slewing is usually caused by internal capacitances in the amplifier, especially those used to implement its frequency compensation.

Non-linear input-output relationship

The output voltage may not be accurately proportional to the difference between the input voltages. It is commonly called distortion when the input signal is a waveform. This effect will be very small in a practical circuit if substantial negative feedback is used.

Power considerations

Limited output current

The output current must be finite. In practice, most op-amps are designed to limit the output current so as not to exceed a specified level — around 25 mA for a type 741 IC op-amp — thus protecting the op-amp and associated circuitry from damage. Modern designs are electronically more rugged than earlier implementations and some can sustain direct short circuits on their outputs without damage.

Limited dissipated power

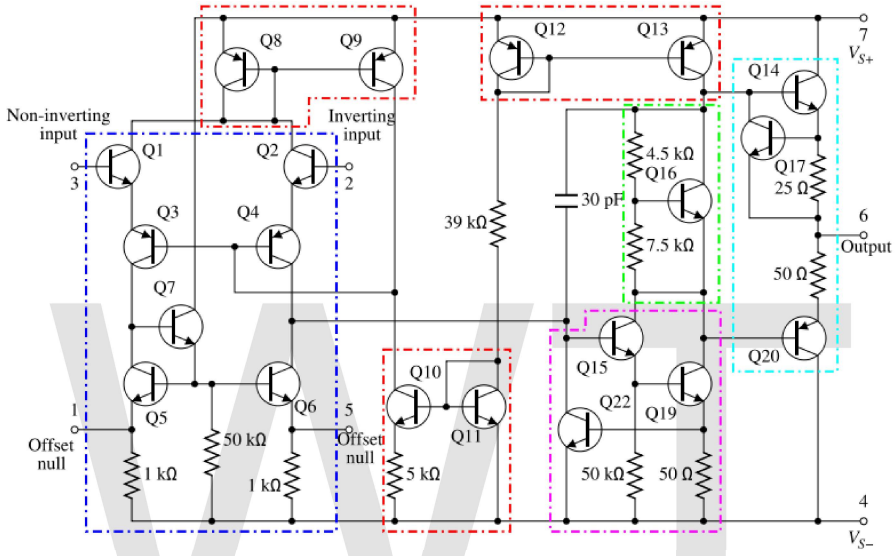
The output current flows through the op-amp's internal output impedance, dissipating heat. If the op-amp dissipates too much power, then its temperature will increase above some safe limit. The op-amp may enter thermal shutdown, or it may be destroyed.

Modern integrated FET or MOSFET op-amps approximate more closely the ideal op-amp than bipolar ICs when it comes to input impedance and input bias and offset

currents. Bipolars are generally better when it comes to input *voltage* offset, and often have lower noise. Generally, at room temperature, with a fairly large signal, and limited bandwidth, FET and MOSFET op-amps now offer better performance.

Internal circuitry of 741 type op-amp

Though designs vary between products and manufacturers, all op-amps have basically the same internal structure, which consists of three stages:



A component level diagram of the common 741 op-amp. Dotted lines outline: current mirrors (red); differential amplifier (blue); class A gain stage (magenta); voltage level shifter (green); output stage (cyan).

1. Differential amplifier – provides low noise amplification, high input impedance, usually a differential output.
2. Voltage amplifier – provides high voltage gain, a single-pole frequency roll-off, usually single-ended output.
3. Output amplifier – provides high current driving capability, low output impedance, current limiting and short circuit protection circuitry.

Input stage

Constant-current stabilization system

The input stage DC conditions are stabilized by a high-gain negative feedback system whose main parts are the two current mirrors on the left of the figure, outlined in red. The main purpose of this negative feedback system—to supply the differential input stage with a stable constant current—is realized as follows.

The current through the 39 kΩ resistor acts as a current reference for the other bias currents used in the chip. The voltage across the resistor is equal to the voltage across the supply rails ($V_{S+} - V_{S-}$) minus two transistor diode drops (i.e., from Q11 and Q12), and so the current has value $I_{\text{ref}} = (V_{S+} - V_{S-} - 2V_{be})/(39\ \text{k}\Omega)$. The Widlar current mirror built by Q10, Q11, and the 5 kΩ resistor produces a very small fraction of I_{ref} at the Q10 collector. This small constant current through Q10's collector supplies the base currents for Q3 and Q4 as well as the Q9 collector current. The Q8/Q9 current mirror tries to make Q9's collector current the same as the Q3 and Q4 *collector* currents. Thus Q3 and Q4's combined base currents (which are of the same order as the overall chip's input currents) will be a small fraction of the already small Q10 current.

So, if the input stage current increases for any reason, the Q8/Q9 current mirror will draw current away from the bases of Q3 and Q4, which reduces the input stage current, and vice versa. The feedback loop also isolates the rest of the circuit from common-mode signals by making the base voltage of Q3/Q4 follow tightly $2V_{be}$ below the higher of the two input voltages.

Differential amplifier

The blue outlined section is a differential amplifier. Q1 and Q2 are input emitter followers and together with the common base pair Q3 and Q4 form the differential input stage. In addition, Q3 and Q4 also act as level shifters and provide voltage gain to drive the class A amplifier. They also help to increase the reverse V_{be} rating on the input transistors (the emitter-base junctions of the NPN transistors Q1 and Q2 break down at around 7 V but the PNP transistors Q3 and Q4 have breakdown voltages around 50 V).

The differential amplifier formed by Q1–Q4 drives a current mirror active load formed by transistors Q5–Q7 (actually, Q6 is the very active load). Q7 increases the accuracy of the current mirror by decreasing the amount of signal current required from Q3 to drive the bases of Q5 and Q6. This configuration provides differential to single ended conversion as follows:

The signal current of Q3 is the input to the current mirror while the output of the mirror (the collector of Q6) is connected to the collector of Q4. Here, the signal currents of Q3 and Q4 are summed. For differential input signals, the signal currents of Q3 and Q4 are equal and opposite. Thus, the sum is twice the individual signal currents. This completes the differential to single ended conversion.

The open circuit signal voltage appearing at this point is given by the product of the summed signal currents and the paralleled collector resistances of Q4 and Q6. Since the collectors of Q4 and Q6 appear as high resistances to the signal current, the open circuit voltage gain of this stage is very high.

The base current at the inputs is not zero and the effective (differential) input impedance of a 741 is about 2 MΩ. The "offset null" pins may be used to place external resistors in parallel with the two 1 kΩ resistors (typically in the form of the two ends of a

potentiometer) to adjust the balancing of the Q5/Q6 current mirror and thus indirectly control the output of the op-amp when zero signal is applied between the inputs.

Class A gain stage

The section outlined in magenta is the class A gain stage. The top-right current mirror Q12/Q13 supplies this stage by a constant current load, via the collector of Q13, that is largely independent of the output voltage. The stage consists of two NPN transistors in a Darlington configuration and uses the output side of a current mirror as its collector load to achieve high gain. The 30 pF capacitor provides frequency selective negative feedback around the class A gain stage as a means of frequency compensation to stabilise the amplifier in feedback configurations. This technique is called Miller compensation and functions in a similar manner to an op-amp integrator circuit. It is also known as 'dominant pole compensation' because it introduces a dominant pole (one which masks the effects of other poles) into the open loop frequency response. This pole can be as low as 10 Hz in a 741 amplifier and it introduces a −3 dB loss into the open loop response at this frequency. This internal compensation is provided to achieve unconditional stability of the amplifier in negative feedback configurations where the feedback network is non-reactive and the closed loop gain is unity or higher. Hence, the use of the operational amplifier is simplified because no external compensation is required for unity gain stability; amplifiers without this internal compensation may require external compensation or closed loop gains significantly higher than unity.

Output bias circuitry

The green outlined section (based on Q16) is a voltage level shifter or rubber diode (i.e., a V_{BE} multiplier); a type of voltage source. In the circuit as shown, Q16 provides a constant voltage drop between its collector and emitter regardless of the current through the circuit. If the base current to the transistor is assumed to be zero, and the voltage between base and emitter (and across the 7.5 kΩ resistor) is 0.625 V (a typical value for a BJT in the active region), then the current through the 4.5 kΩ resistor will be the same as that through the 7.5 kΩ, and will produce a voltage of 0.375 V across it. This keeps the voltage across the transistor, and the two resistors at 0.625 + 0.375 = 1 V. This serves to bias the two output transistors slightly into conduction reducing crossover distortion. In some discrete component amplifiers this function is achieved with (usually two) silicon diodes.

Output stage

The output stage (outlined in cyan) is a Class AB push-pull emitter follower (Q14, Q20) amplifier with the bias set by the V_{be} multiplier voltage source Q16 and its base resistors. This stage is effectively driven by the collectors of Q13 and Q19. Variations in the bias with temperature, or between parts with the same type number, are common so crossover distortion and quiescent current may be subject to significant variation. The output range of the amplifier is about one volt less than the supply voltage, owing in part to V_{be} of the output transistors Q14 and Q20.

The 25 Ω resistor in the output stage acts as a current sense to provide the output current-limiting function which limits the current in the emitter follower Q14 to about 25 mA for the 741. Current limiting for the negative output is done by sensing the voltage across Q19's emitter resistor and using this to reduce the drive into Q15's base. Later versions of this amplifier schematic may show a slightly different method of output current limiting. The output resistance is not zero, as it would be in an ideal op-amp, but with negative feedback it approaches zero at low frequencies.

Note: while the 741 was historically used in audio and other sensitive equipment, such use is now rare because of the improved noise performance of more modern op-amps. Apart from generating noticeable hiss, 741s and other older op-amps may have poor common-mode rejection ratios and so will often introduce cable-borne mains hum and other common-mode interference, such as switch 'clicks', into sensitive equipment.

The "741" has come to often mean a generic op-amp IC (such as uA741, LM301, 558, LM324, TBA221 - or a more modern replacement such as the TL071). The description of the 741 output stage is qualitatively similar for many other designs (that may have quite different input stages), except:

- Some devices (uA748, LM301, LM308) are not internally compensated (require an external capacitor from output to some point within the operational amplifier, if used in low closed-loop gain applications).
- Some modern devices have rail-to-rail output capability (output can be taken to positive or negative power supply rail within a few millivolts).

Chapter 11

Operational Amplifier Applications

Practical considerations

Input offset problems

It is important to note that the equations shown below, pertaining to each type of circuit, assume that an ideal op amp is used.

Resistors used in practical solid-state op-amp circuits are typically in the kΩ range. Resistors much greater than 1 MΩ cause excessive thermal noise and make the circuit operation susceptible to significant errors due to bias or leakage currents.

Practical operational amplifiers draw a small current from each of their inputs due to bias requirements and leakage. These currents flow through the resistances connected to the inputs and produce small voltage drops across those resistances. In AC signal applications this seldom matters. If high-precision DC operation is required, however, these voltage drops need to be considered. The design technique is to try to ensure that these voltage drops are equal for both inputs, and therefore cancel. If these voltage drops are equal and the common-mode rejection ratio of the operational amplifier is good, there will be considerable cancellation and improvement in DC accuracy.

If the input currents into the operational amplifier are equal, to reduce offset voltage the designer must ensure that the DC resistance looking out of each input is also matched. In general input currents differ, the difference being called the *input offset current*, I_{os}. Matched external input resistances R_{in} will still produce an input voltage error of $R_{in} \cdot I_{os}$. Most manufacturers provide a method for tuning the operational amplifier to balance the input currents (e.g., "offset null" or "balance" pins that can interact with an external voltage source attached to a potentiometer). Otherwise, a tunable external voltage can be added to one of the inputs in order to balance out the offset effect. In cases where a design calls for one input to be short-circuited to ground, that short circuit can be replaced with a variable resistance that can be tuned to mitigate the offset problem.

Note that many operational amplifiers that have MOSFET-based input stages have input leakage currents that will truly be negligible to most designs.

Power supply effects

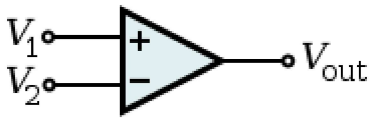
Although the power supplies are not shown in the operational amplifier designs below, they can be critical in operational amplifier design.

Power supply imperfections (e.g., power signal ripple, non-zero source impedance) may lead to noticeable deviations from ideal operational amplifier behavior. For example, operational amplifiers have a specified power supply rejection ratio that indicates how well the output can reject signals that appear on the power supply inputs. Power supply inputs are often noisy in large designs because the power supply is used by nearly every component in the design, and inductance effects prevent current from being instantaneously delivered to every component at once. As a consequence, when a component requires large injections of current (e.g., a digital component that is frequently switching from one state to another), nearby components can experience sagging at their connection to the power supply. This problem can be mitigated with copious use of bypass capacitors placed connected across each power supply pin and ground. When bursts of current are required by a component, the component can *bypass* the power supply by receiving the current directly from the nearby capacitor (which is then slowly charged by the power supply).

Additionally, current drawn into the operational amplifier from the power supply can be used as inputs to external circuitry that augment the capabilities of the operational amplifier. For example, an operational amplifier may not be fit for a particular high-gain application because its output would be required to generate signals outside of the safe range generated by the amplifier. In this case, an external push–pull amplifier can be controlled by the current into and out of the operational amplifier. Thus, the operational amplifier may itself operate within its factory specified bounds while still allowing the negative feedback path to include a large output signal well outside of those bounds.

Circuit applications

Comparator

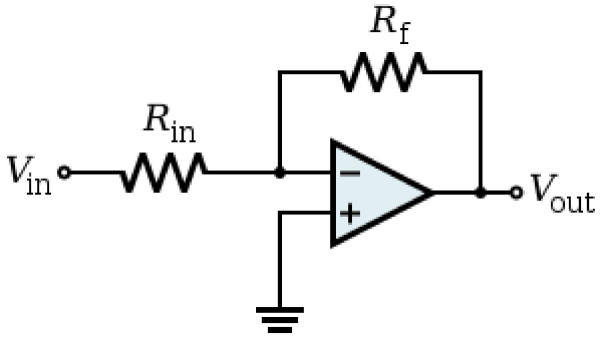


Compares two voltages and switches its output to indicate which voltage is larger.

- $$V_{\text{out}} = \begin{cases} V_{\text{S+}} & V_1 > V_2 \\ V_{\text{S-}} & V_1 < V_2 \end{cases}$$

(where V_s is the supply voltage and the opamp is powered by $+V_s$ and $-V_s$.)

Inverting amplifier



An inverting amplifier uses negative feedback to invert and amplify a voltage. The R_{in},R_f resistor network allows some of the output signal to be returned to the input. Since the output is 180° out of phase, this amount is effectively subtracted from the input, thereby reducing the input into the operational amplifier. This reduces the overall gain of the amplifier and is dubbed negative feedback.

$$V_{\text{out}} = -\frac{R_{\text{f}}}{R_{\text{in}}} V_{\text{in}}$$

- $Z_{\text{in}} = R_{\text{in}}$ (because V_- is a virtual ground)
- A third resistor, of value $R_{\text{f}} \| R_{\text{in}} \triangleq R_{\text{f}} R_{\text{in}} / (R_{\text{f}} + R_{\text{in}})$, added between the non-inverting input and ground, while not necessary, minimizes errors due to input bias currents.

The gain of the amplifier is determined by the ratio of R_f to R_{in}. That is:

$$A = -\frac{R_f}{R_{in}}$$

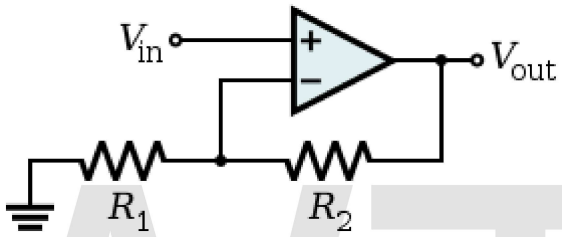
The presence of the negative sign is a convention indicating that the output is inverted. For example, if R_f is 10 000 Ω and R_{in} is 1 000 Ω, then the gain would be -10 000Ω/1 000Ω, which is -10.

Theory of operation: An Ideal Operational Amplifier has 2 characteristics that imply the operation of the inverting amplifier: Infinite input impedance, and infinite differential gain. Infinite input impedance implies there is no current in either of the input pins because current cannot flow through an infinite impedance. Infinite differential gain implies that both the (+) and (-) input pins are at the same voltage because the output is equal to infinity times (V+ - V-). As the output approaches any arbitrary finite voltage, then the term (V+ - V-) approaches 0, thus the two input pins are at the same voltage for any finite output.

To begin analysis, first it is noted that with the (+) pin grounded, the (-) must also be at 0 volts potential due to implication 2. with the (-) at 0 volts, the current through Rin (from left to right) is given by I = Vin/Rin by Ohm's law. Second, since no current is flowing into the op amp through the (-) pin due to implication 1, all the current through Rin must also be flowing through Rf. Therefore, with V- = 0 volts and I(Rf) = Vin/Rin the output voltage given by Ohm's law is -Vin*Rf/Rin.

Real op amps have both finite input impedance and differential gain, however both are high enough as to induce error that is considered negligible in most applications.

Non-inverting amplifier



Amplifies a voltage (multiplies by a constant greater than 1)

$$V_{\text{out}} = V_{\text{in}} \left(1 + \frac{R_2}{R_1}\right)$$

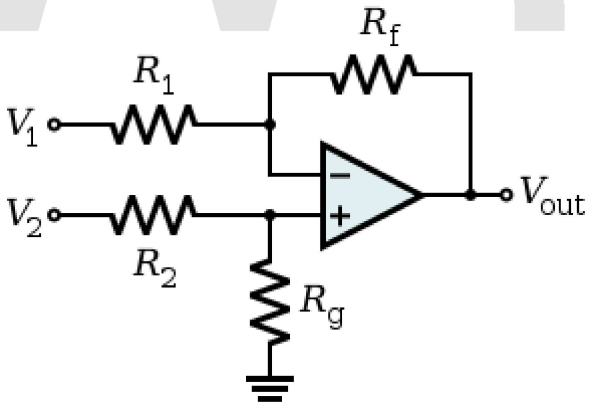
- Input impedance $Z_{\text{in}} \approx \infty$
  - The input impedance is *at least* the impedance between non-inverting (+) and inverting (−) inputs, which is typically 1 MΩ to 10 TΩ, plus the impedance of the path from the inverting (−) input to ground (i.e., R_1 in parallel with R_2).
  - Because negative feedback ensures that the non-inverting and inverting inputs match, the input impedance is actually **much higher.**
- Although this circuit has a large input impedance, it suffers from error of input bias current.
  - The non-inverting (+) and inverting (−) inputs draw small leakage currents into the operational amplifier.
  - These input currents generate voltages that act like unmodeled input offsets. These unmodeled effects can lead to noise on the output (e.g., offsets or drift).
  - Assuming that the two leaking currents are **matched,** their effect can be mitigated by ensuring the DC impedance looking **out** of each input is the same.
    - The voltage produced by each bias current is equal to the product of the bias current with the equivalent DC impedance looking out of each input. Making those impedances equal makes the offset

voltage at each input equal, and so the non-zero bias currents will have no impact on the **difference** between the two inputs.
        - A resistor of value

$$R_1 \| R_2 \triangleq \left(\frac{1}{R_1} + \frac{1}{R_2} \right)^{-1} = \frac{R_1 R_2}{R_1 + R_2},$$

    - which is the equivalent resistance of R_1 in parallel with R_2, between the V_{in} source and the non-inverting (+) input will ensure the impedances looking **out** of each input will be matched.
        - The matched bias currents will then generate matched offset voltages, and their effect will be hidden to the operational amplifier (which acts on the difference between its inputs) so long as the CMRR is good.
    - Very often, the input currents are *not* matched.
        - Most operational amplifiers provide some method of balancing the two input currents (e.g., by way of an external potentiometer).
        - Alternatively, an external offset can be added to the operational amplifier input to nullify the effect.
        - Another solution is to insert a variable resistor between the V_{in} source and the non-inverting (+) input. The resistance can be tuned until the offset voltages at each input are matched.
        - Operational amplifiers with MOSFET-based input stages have input currents that are so small that they often can be neglected.

Differential amplifier



The circuit shown is used for finding the difference of two voltages each multiplied by some constant (determined by the resistors).

$$V_{out} = \frac{(R_f + R_1) R_g}{(R_g + R_2) R_1} V_2 - \frac{R_f}{R_1} V_1$$

- Differential Z_{in} (between the two input pins) = $R_1 + R_2$ (Note: this is approximate)

For common-mode rejection, anything done to one input must be done to the other. The addition of a compensation capacitor in parallel with Rf, for instance, must be balanced by an equivalent capacitor in parallel with Rg.

The "instrumentation amplifier", which is also shown on this page, is another form of differential amplifier that also provides high input impedance.

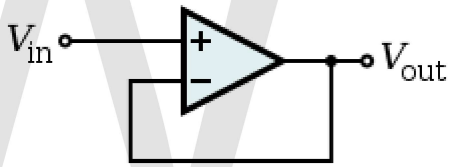
Whenever $R_1 = R_2$and $R_f = R_g$, the differential gain is

$$V_{out} = A(V_2 - V_1) \text{ and } A \triangleq \frac{R_f}{R_1}$$

When $R_1 = R_f$and $R_2 = R_g$the differential gain is A = 1 and the circuit acts as a differential follower:

$$V_{out} = V_2 - V_1$$

Voltage follower

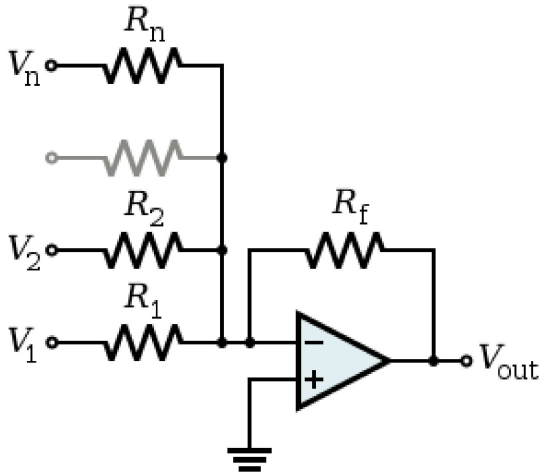


Used as a buffer amplifier to eliminate loading effects (e.g., connecting a device with a high source impedance to a device with a low input impedance).

$V_{out} = V_{in}$
$Z_{in} = \infty$(realistically, the differential input impedance of the op-amp itself, 1 MΩ to 1 TΩ)

Due to the strong (i.e., unity gain) feedback and certain non-ideal characteristics of real operational amplifiers, this feedback system is prone to have poor stability margins. Consequently, the system may be unstable when connected to sufficiently capacitive loads. In these cases, a lag compensation network (e.g., connecting the load to the voltage follower through a resistor) can be used to restore stability. The manufacturer data sheet for the operational amplifier may provide guidance for the selection of components in external compensation networks. Alternatively, another operational amplifier can be chosen that has more appropriate internal compensation.

Summing amplifier



A summing amplifer sums several (weighted) voltages:

$$V_{\text{out}} = -R_{\text{f}} \left(\frac{V_1}{R_1} + \frac{V_2}{R_2} + \cdots + \frac{V_n}{R_n} \right)$$

- When $R_1 = R_2 = \cdots = R_n$, and R_{f} independent

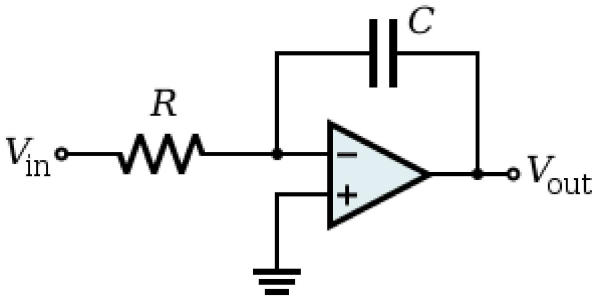
$$V_{\text{out}} = -\frac{R_{\text{f}}}{R_1}(V_1 + V_2 + \cdots + V_n)$$

- When $R_1 = R_2 = \cdots = R_n = R_{\text{f}}$

$$V_{\text{out}} = -(V_1 + V_2 + \cdots + V_n)$$

- Output is inverted
- Input impedance of the ***n*th** input is $Z_n = R_n$ (V_- is a virtual ground)

Inverting integrator



Integrates the (inverted) signal over time

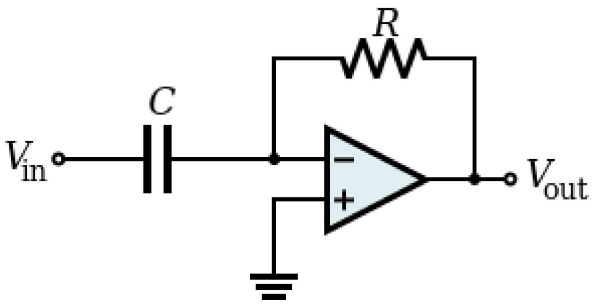
$$V_{\text{out}} = -\int_0^t \frac{V_{\text{in}}}{RC}\,\mathrm{d}t + V_{\text{initial}}$$

(where V_{in} and V_{out} are functions of time, V_{initial} is the output voltage of the integrator at time $t = 0$.)

- Note that this can also be viewed as a low-pass electronic filter. It is a filter with a single pole at DC (i.e., where ω = 0) and gain.
- There are several potential problems with this circuit.
  - It is usually assumed that the input V_{in} has zero DC component (i.e., has a zero average value). Otherwise, unless the capacitor is periodically discharged, the output will drift outside of the operational amplifier's operating range.
  - Even when V_{in} has no offset, the leakage or bias currents into the operational amplifier inputs can add an unexpected offset voltage to V_{in} that causes the output to drift. Balancing input currents **and** replacing the non-inverting (+) short-circuit to ground with a resistor with resistance R can reduce the severity of this problem.
  - Because this circuit provides no DC feedback (i.e., the capacitor appears like an open circuit to signals with ω = 0), the offset of the output may not agree with expectations (i.e., V_{initial} may be out of the designer's control with the present circuit).

  Many of these problems can be made less severe by adding a *large* resistor R_F in parallel with the feedback capacitor. At significantly high frequencies, this resistor will have negligible effect. However, at low frequencies where there are drift and offset problems, the resistor provides the necessary feedback to hold the output steady at the correct value. In effect, this resistor reduces the DC gain of the "integrator" – it goes from infinite to some finite value R_F / R.

Inverting differentiator

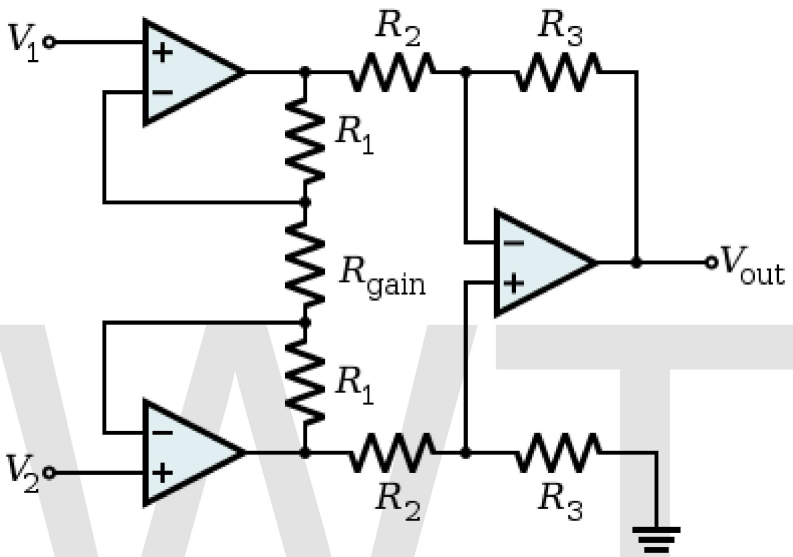


Differentiates the (inverted) signal over time.

$$V_{\text{out}} = -RC\,\frac{\mathrm{d}V_{\text{in}}}{\mathrm{d}t} \qquad \text{where } V_{\text{in}} \text{ and } V_{\text{out}} \text{ are functions of time.}$$

- Note that this can also be viewed as a high-pass electronic filter. It is a filter with a single zero at DC (i.e., where ω = 0) and gain. The high pass characteristics of a differentiating amplifier can lead to unstable behavior when the circuit is used in an analog servo loop. For this reason the system function would be re-formulated to use integrators.

Instrumentation amplifier

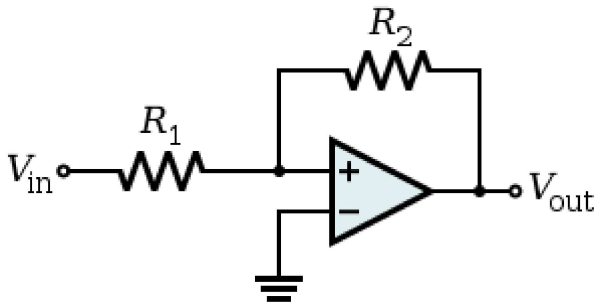


Combines very high input impedance, high common-mode rejection, low DC offset, and other properties used in making very accurate, low-noise measurements

- Is made by adding a non-inverting buffer to each input of the differential amplifier to increase the input impedance.

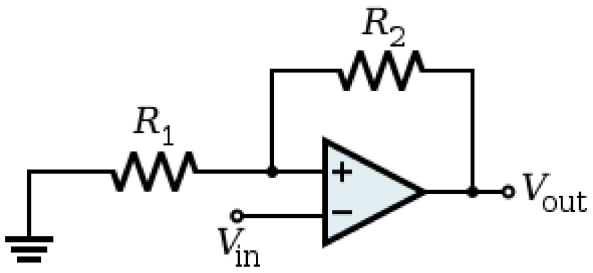
Schmitt trigger

A bistable multivibrator implemented as a comparator with hysteresis.



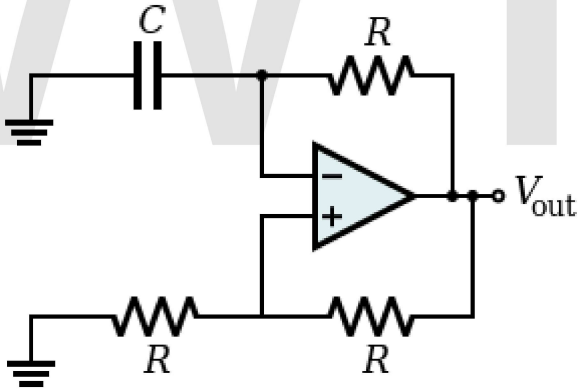
In this configuration, the input voltage is applied through the resistor R_1 (which may be the source internal resistance) to the non-inverting input and the inverting input is

grounded or referenced. The hysteresis curve is non-inverting and the switching thresholds are $\pm\frac{R_1}{R_2}V_{\mathrm{sat}}$ where V_{sat} is the greatest output magnitude of the operational amplifier.



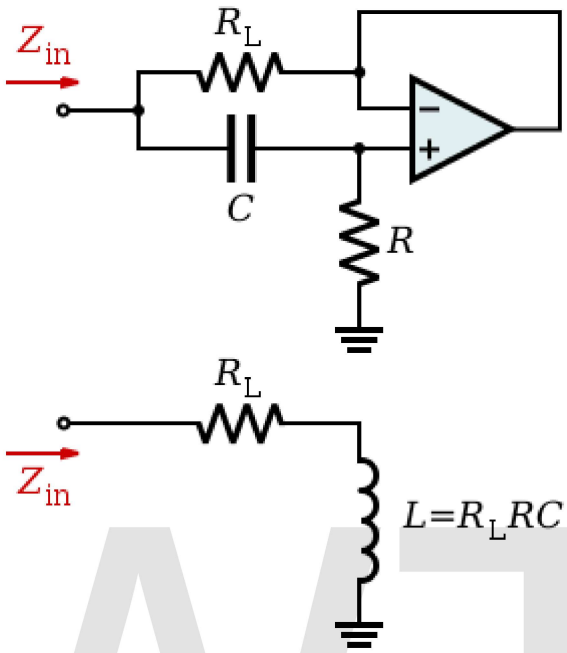
Alternatively, the input source and the ground may be swapped. Now the input voltage is applied directly to the inverting input and the non-inverting input is grounded or referenced. The hysteresis curve is inverting and the switching thresholds are $\pm\frac{R_1}{R_1+R_2}V_{\mathrm{sat}}$. Such a configuration is used in the relaxation oscillator shown below.

Relaxation oscillator



By using an RC network to add slow negative feedback to the inverting Schmitt trigger, a relaxation oscillator is formed. The feedback through the RC network causes the Schmitt trigger output to oscillate in an endless symmetric square wave (i.e., the Schmitt trigger in this configuration is an astable multivibrator).

Inductance gyrator



Simulates an inductor (i.e., provides inductance without the use of a possibly costly inductor). The circuit exploits the fact that the current flowing through a capacitor behaves through time as the voltage across an inductor. The capacitor used in this circuit is smaller than the inductor it simulates and its capacitance is less subject to changes in value due to environmental changes.

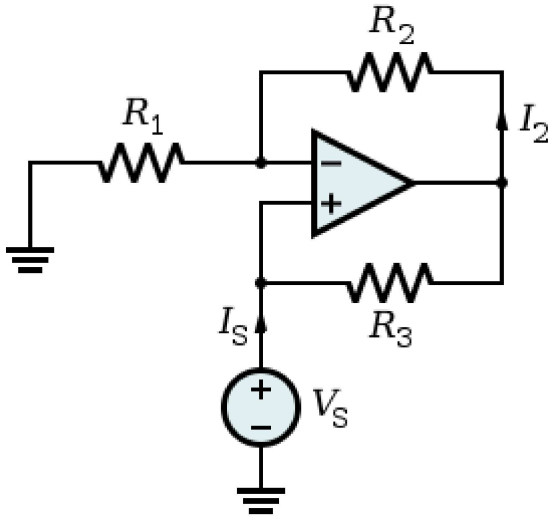
This circuit is unsuitable for applications relying on the back EMF property of an inductor as this will be limited in a gyrator circuit to the voltage supplies of the op-amp.

Zero level detector

Voltage divider reference

- Zener sets reference voltage.
- Acts as a comparator with one input tied to ground.
- When input is at zero, op-amp output is zero (assuming split supplies.)

Negative impedance converter (NIC)



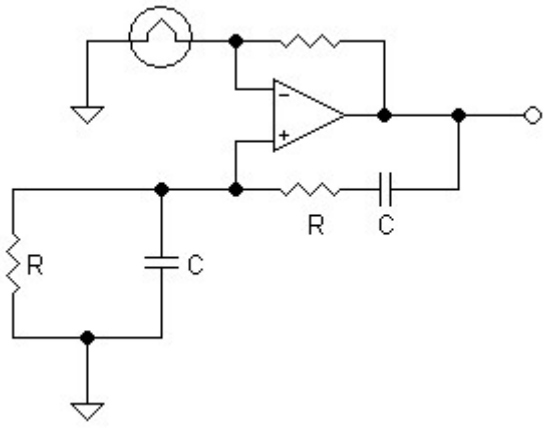
Creates a resistor having a negative value for any signal generator

- In this case, the ratio between the input voltage and the input current (thus the input resistance) is given by:

$$R_{\text{in}} = -R_3 \frac{R_1}{R_2}$$

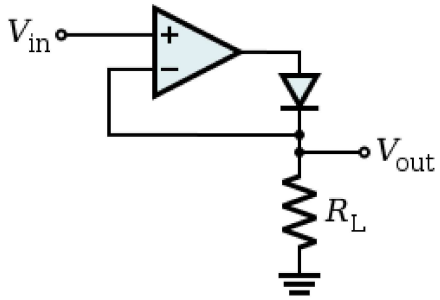
In general, the components R_1, R_2, and R_3 need not be resistors; they can be any component that can be described with an impedance.

Wien bridge oscillator



Produces a very low distortion sine wave. Uses negative temperature compensation in the form of a light bulb or diode.

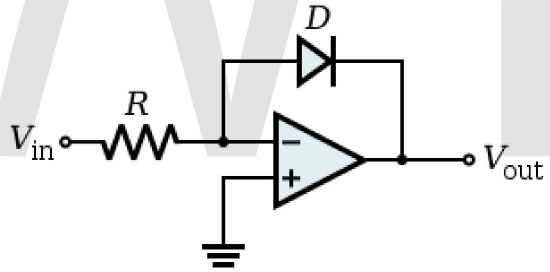
Precision rectifier



The voltage drop V_F across the forward biased diode in the circuit of a passive rectifier is undesired. In this active version, the problem is solved by connecting the diode in the negative feedback loop. The op-amp compares the output voltage across the load with the input voltage and increases its own output voltage with the value of V_F. As a result, the voltage drop V_F is compensated and the circuit behaves very nearly as an ideal (*super*) diode with $V_F = 0$ V.

The circuit has speed limitations at high frequency because of the slow negative feedback and due to the low slew rate of many non-ideal op-amps.

Logarithmic output



- The relationship between the input voltage v_{in} and the output voltage v_{out} is given by:

$$v_{\mathrm{out}} = -V_{\mathrm{T}} \ln\left(\frac{v_{\mathrm{in}}}{I_{\mathrm{S}} R}\right)$$

where I_S is the *saturation current* and V_T is the *thermal voltage*.

- If the operational amplifier is considered ideal, the negative pin is virtually grounded, so the current flowing into the resistor from the source (and thus through the diode to the output, since the op-amp inputs draw no current) is:

$$\frac{v_{\mathrm{in}}}{R} = I_{\mathrm{R}} = I_{\mathrm{D}}$$

where I_D is the current through the diode. As known, the relationship between the current and the voltage for a diode is:

$$I_D = I_S \left(e^{\frac{V_D}{V_T}} - 1 \right).$$

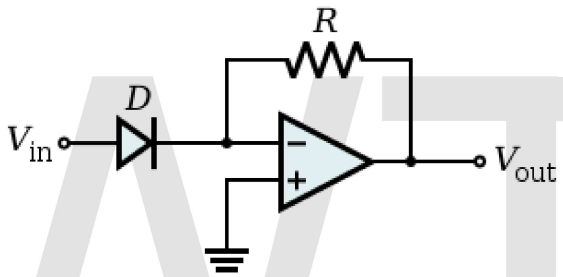
This, when the voltage is greater than zero, can be approximated by:

$$I_D \simeq I_S e^{\frac{V_D}{V_T}}.$$

Putting these two formulae together and considering that the output voltage is the negative of the voltage across the diode ($V_{out} = -V_D$), the relationship is proven.

Note that this implementation does not consider temperature stability and other non-ideal effects.

Exponential output



- The relationship between the input voltage v_{in} and the output voltage v_{out} is given by:

$$v_{out} = -RI_S e^{\frac{v_{in}}{V_T}}$$

where I_S is the *saturation current* and V_T is the *thermal voltage*.

- Considering the operational amplifier ideal, then the negative pin is virtually grounded, so the current through the diode is given by:

$$I_D = I_S \left(e^{\frac{V_D}{V_T}} - 1 \right)$$

when the voltage is greater than zero, it can be approximated by:

$$I_D \simeq I_S e^{\frac{V_D}{V_T}}.$$

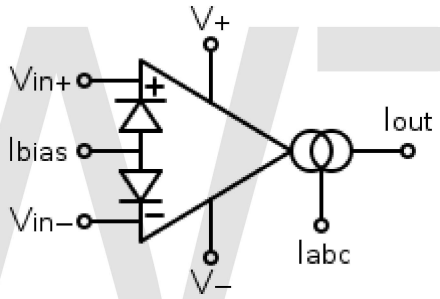
The output voltage is given by:

$$v_{out} = -RI_D.$$

Chapter 12

Operational Transconductance Amplifier & Low-Dropout Regulator

Operational Transconductance Amplifier



Schematic symbol for the OTA. Like the standard operational amplifier, it has both inverting (−) and noninverting (+) inputs; power supply lines (V+ and V−); and a single output. Unlike the traditional op-amp, it has two additional biasing inputs, I_{abc} and I_{bias}, explained in **Basic operation** and **Subsequent improvements**, below.

The **operational transconductance amplifier** (**OTA**) is an amplifier whose differential input voltage produces an output current. Thus, it is a voltage controlled current source (VCCS). There is usually an additional input for a current to control the amplifier's transconductance. The OTA is similar to a standard operational amplifier in that it has a high impedance differential input stage and that it may be used with negative feedback.

The first commercially available integrated circuit units were produced by RCA in 1969 (before being acquired by General Electric), in the form of the CA3080, and they have been improved since that time. Although most units are constructed with bipolar transistors, field effect transistor units are also produced. The OTA is not as useful by itself in the vast majority of standard op-amp functions as the ordinary op-amp because its output is a current. One of its principal uses is in implementing electronically controlled applications such as variable frequency oscillators and filters and variable gain amplifier stages which are more difficult to implement with standard op-amps.

Principal differences from standard operational amplifiers

- Its output of a *current* contrasts to that of standard operational amplifier whose output is a *voltage*.

- It is usually used "open-loop"; without negative feedback in linear applications. This is possible because the magnitude of the resistance attached to its output controls its output voltage. Therefore a resistance can be chosen that keeps the output from going into saturation, even with high differential input voltages.

Basic operation

In the ideal OTA, the output current is a linear function of the differential input voltage, calculated as follows:

$$I_{\text{out}} = (V_{\text{in}+} - V_{\text{in}-}) \cdot g_{\text{m}}$$

where $V_{\text{in}+}$ is the voltage at the non-inverting input, $V_{\text{in}-}$ is the voltage at the inverting input and g_m is the transconductance of the amplifier.

The amplifier's output voltage is the product of its output current and its load resistance:

$$V_{\text{out}} = I_{\text{out}} \cdot R_{\text{load}}$$

The voltage gain is then the output voltage divided by the differential input voltage:

$$G_{\text{voltage}} = \frac{V_{\text{out}}}{(V_{\text{in}+} - V_{\text{in}-})} = R_{\text{load}} \cdot g_{\text{m}}$$

The transconductance of the amplifier is usually controlled by an input current, denoted I_{abc} ("amplifier bias current"). The amplifier's transconductance is directly proportional to this current. This is the feature that makes it useful for electronic control of amplifier gain, etc.

Non-ideal characteristics

As with the standard op-amp, practical OTA's have some non-ideal characteristics. These include:

- Input stage non-linearity at higher differential input voltages due to the characteristics of the input stage transistors. In the early devices, such as the CA 3080, the input stage consisted of two bipolar transistors connected in the differential amplifier configuration. The transfer characteristics of this connection are approximately linear for differential input voltages of 20 mV or less. This is an important limitation when the OTA is being used open loop as there is no

negative feedback to linearize the output. One scheme to improve this parameter is mentioned below.

- Temperature sensitivity of transconductance.
- Variation of input and output impedance, input bias current and input offset voltage with the transconductance control current I_{abc}.

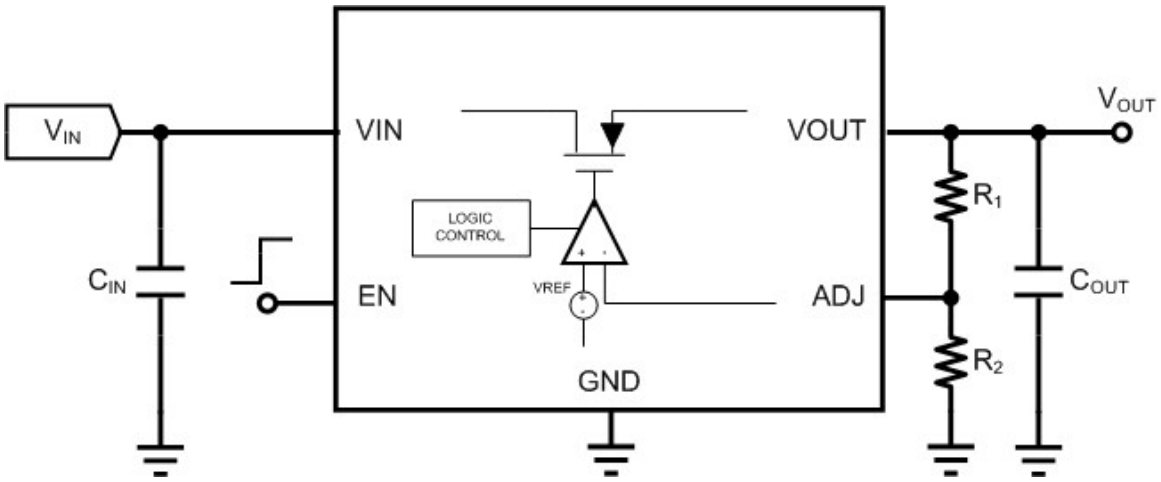
Subsequent improvements

Earlier versions of the OTA had neither the I_{bias} terminal shown in the diagram nor the diodes shown adjacent to it. They were all added in later versions. As depicted in the diagram, the anodes of the diodes are attached together and the cathode of one is attached to the non inverting input (Vin+) and the cathode of the other to the inverting input (Vin−). The diodes are biased at the anodes by a current (I_{bias}) that is injected into the I_{bias} terminal. These additions make two substantial improvements to the OTA. First, when used with input resistors, the diodes distort the differential input voltage to offset a significant amount of input stage non linearity at higher differential input voltages. According to National Semiconductor, the addition of these diodes increases the linearity of the input stage by a factor of 4. That is, using the diodes, the signal distortion level at 80 mV of differential input is the same as that of the simple differential amplifier at a differential input of 20 mV. Second, the action of the biased diodes offsets much of the temperature sensitivity of the OTA's transconductance.

A second improvement is the integration of an optional-use output buffer amplifier to the chip on which the OTA resides. This is actually a convenience to a circuit designer rather than an improvement to the OTA itself; dispensing with the need to employ a separate buffer. It also allows the OTA to be used as a traditional op-amp, if desired, by converting its output current to a voltage.

An example of a chip combining both of these features is the National Semiconductor LM13600 and its successor, the LM13700, the data sheet for which can be found here:

Low-Dropout Regulator



Schematic of a low-dropout regulator

A **low-dropout** or **LDO** regulator is a DC linear voltage regulator which can operate with a very small input–output differential voltage. The advantages of a low dropout voltage include a lower minimum operating voltage, higher efficiency operation and lower heat dissipation.

The main components are a power FET and a differential amplifier (error amplifier). One input of the differential amplifier monitors the fraction of the output determined by the resistor ratio of **R1** and **R2**. The second input to the differential amplifier is from a stable voltage reference (bandgap reference). If the output voltage rises too high relative to the reference voltage, the drive to the power FET changes to maintain a constant output voltage.

Overview

The adjustable low-dropout regulator debuted on April 12, 1977 in an Electronic Design article entitled "*Break Loose from Fixed IC Regulators*". The article was written by Robert Dobkin, an IC designer then working for National Semiconductor. Because of this, National Semiconductor claims the title of "*LDO inventor*". Dobkin later left National in 1981 to found Linear Technology where he is currently chief technology officer.

Current advances in semiconductor technology made possible to significantly reduce voltage drop on all linear voltage regulators, so producers today advertise almost everything as low-dropout. For example, the AS1117 is advertised as low drop out even if it uses same topology as 78xx series.

Regulation

Low-dropout (LDO) regulators work in the same way as all linear voltage regulators. The main difference between LDO and non-LDO regulators is their schematic topology. Instead of an emitter follower topology, Low-dropout regulators utilize open collector or open drain topology. This enables transistor saturation, which limits the voltage drop to only the saturation voltage.

If a bipolar transistor is used however, significant additional power is lost to control it, when non-LDO regulators take that power from voltage drop itself. For high voltages under very low In-Out difference there will be significant power loss in control circuit.

Because the power control element is working as an inverter, another inverting amplifier is required to control it, which increases schematic complexity compared to a simple voltage stabilizer.

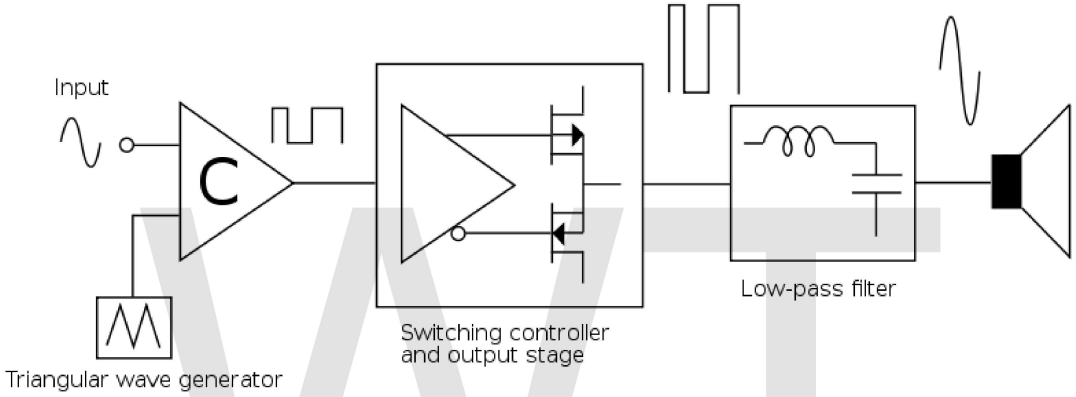
Power FETs may be preferable to reduce power consumption, but this significantly increases price and poses problems when the regulator is used for low input voltage, because FETs usually require 10 to 5V to open completely.

Quiescent current

Among other important characteristics is the quiescent current (the current flowing through the system when no load is present), which creates a difference between the input and output currents. The series pass element, topologies, and ambient temperature are the primary contributors to quiescent current. Quiescent current and input–output drop limit the efficiency of LDO regulators and should thus be minimized.

Chapter 13

Class D Amplifier



Block diagram of a basic switching or PWM (class D) amplifier

A **Class D amplifier** or **switching amplifier** is an electronic amplifier where all power devices (usually MOSFETs) are operated as binary switches. They are either fully on or fully off. Ideally, zero time is spent transitioning between those two states.

Output stages such as those used in pulse generators are examples of class D amplifiers. However, the term mostly applies to power amplifiers intended to reproduce signals with a bandwidth well below the switching frequency.

Basic operation

Class D amplifiers work by generating a square wave of which the low-frequency portion of the spectrum is essentially the wanted output signal, and of which the high-frequency portion serves no purpose other than to make the wave-form binary so it can be amplified by switching the power devices.

A passive low-pass filter removes the unwanted high-frequency components, ie. smooths the pulses out and recovers the desired low-frequency signal. To maintain high efficiency, the filter is made with purely reactive components (inductors and capacitors), which store the excess energy until it is needed instead of converting some of it into heat. The switching frequency is typically chosen to be ten or more times the highest frequency of interest in the input signal. This eases the requirements placed on the output filter.

The structure of a class D power stage is essentially identical to that of a synchronously rectified buck converter, a type of non-isolated switched-mode power supply. Whereas buck converters usually function as voltage regulators, delivering a constant DC voltage into a variable load and can only source current (one-quadrant operation), a class D amplifier delivers a constantly changing voltage into a fixed load, where current and voltage can independently change sign (four-quadrant operation). A switching amplifier must not be confused with any amplifier that uses an SMPS. A switching amplifier may use any type of power supply but the amplification process itself operates by switching.

Theoretical power efficiency of class D amplifiers is 100%. That is to say, all of the power supplied to it is delivered to the load, none is turned to heat. This is because an ideal switch in its on state will conduct all current but has no voltage across it, hence no heat is dissipated. And when it is off, it will have the full supply voltage standing across it, but no current flows through it. Again, no heat is dissipated. Real-life power MOSFETs are not ideal switches, but practical efficiencies well over 90% are common. By contrast, linear AB-class amplifiers are always operated with both current flowing through and voltage standing across the power devices. An ideal class B amplifier has a theoretical maximum efficiency of 78%.

Terminology

The term "class D" is sometimes misunderstood as meaning a "digital" amplifier. While some class D amps may indeed be controlled by digital circuits, the power stage deals with voltage and current as a function of time. The smallest amount of noise, timing uncertainty, voltage ripple or any other non-ideality immediately results in an irreversible change of the output signal. A digital circuit also uses physics to operate, but those same errors will only lead to incorrect results when they become so large that a signal representing a digit is distorted beyond recognition. Up to that point, non-idealities have no impact on the transmitted signal. The difference between digital and analogue signals is that digital signals are subsequently interpreted as numbers whereas in analogue signals the exact waveform matters.

Nevertheless, the term "digital amplifier" has gained currency to denote class D amplifiers with significant amounts of digital processing in them.

Signal modulation

The binary waveform is derived using pulse-width modulation (PWM), pulse density modulation (sometimes referred to as pulse frequency modulation), sliding mode control (more commonly called "self-oscillating modulation" in the trade) or discrete-time forms of modulation such as delta-sigma modulation.

The most basic way of creating the PWM signal is to use a high speed comparator ("**C**" in the block-diagram above) that compares a high frequency triangular wave with the audio input. This generates a series of pulses of which the duty cycle is directly proportional with the instantaneous value of the audio signal. The comparator then drives

a MOS gate driver which in turn drives a pair of high-power swiitches (usually MOSFETs). This produces an amplified replica of the comparator's PWM signal. The output filter removes the high-frequency switching components of the PWM signal and recovers the audio information that the speaker can use.

DSP-based amplifiers which generate a PWM signal directly from a digital audio signal (e.g. SPDIF) either use a counter to time the pulse length e.g. or implement a digital equivalent of a triangle-based modulator. In either case, the time resolution afforded by practical clock frequencies is only a few hundredths of a switching period, which is not enough to insure low noise. In effect, the pulse length gets quantized, resulting in quantization distortion. In both cases, negative feedback is applied inside the digital domain, forming a noise shaper which has lower noise in the audible frequency range.

Design Challenges

Two significant design challenges for MOSFET driver circuits in class D amplifiers are keeping dead times and linear mode operation as short as possible. "Dead time" is the period during a switching transition when both output MOSFETs are driven into Cut-Off Mode and both are "off". Dead times need to be as short as possible to maintain an accurate low-distortion output signal, but dead times that are too short cause the MOSFET that is switching on to start conducting before the MOSFET that is switching off has stopped conducting. The MOSFETs effectively short the output power supply through themselves, a condition known as "shoot-through". Meanwhile, the MOSFET drivers also need to drive the MOSFETs between switching states as fast as possible to minimize the amount of time a MOSFET is in Linear Mode, the state between Cut-Off Mode and Saturation Mode where the MOSFET is neither fully on nor fully off and conducts current with a significant resistance, creating significant heat. Driver failures that allow shoot-through and/or too much linear mode operation result in excessive losses and sometimes catastrophic failure of the MOSFETs .

Error Control

The actual output of the amplifier is not just dependent on the content of the modulated PWM signal. The power supply voltage directly amplitude-modulates the output voltage, dead time errors make the output impedance non-linear and the output filter has a strongly load-dependent frequency response. An effective way to combat errors, regardless of their source, is negative feedback. A feedback loop including the output stage can be made using a simple integrator. To include the output filter, a PID controller is used, sometimes with additional integrating terms. The need to feed the actual output signal back into the modulator makes the direct generation of PWM from a SPDIF source unattractive .

Advantages

Despite the complexity involved, a properly designed class D amplifier offers the following benefits:

- Reduction in size and weight of the amplifier,
- Reduced power waste as heat dissipation and hence smaller (or no) heat sinks,
- Reduction in cost due to smaller heat sink and compact circuitry,
- Very high power conversion efficiency, usually above 90% above one quarter of the amplifier's maximum power, and around 50% at low power levels.

Boss Audio mono amp. The output stage is top left, the output chokes are the two yellow toroids underneath.

Uses

- Home Theatre systems. In particular the economical "home theatre in a box" systems are almost universally equipped with class D amplifiers. On account of modest performance requirements and straightforward design, direct conversion from digital audio to PWM without feedback is most common.
- Mobile phones. The internal loudspeaker is driven by up to 1 W. Class D is used to preserve battery lifetime.
- Powered speakers
- High-end audio is generally conservative with regards to adopting new technologies but class D amplifiers have made an appearance
- Active subwoofers

- Sound Reinforcement and Live Sound. The weight reduction makes class D amplifiers more transportable. The Crest Audio CD3000, for example, is a class D power amplifier that is rated at 1500 W per channel, yet it weighs only 21 kg (46 lb).
- Bass amplifiers Again, an area where portability is important. Example: Yamaha BBT500H bass amplifier which is rated at 500 W, and yet it weighs less than 5 kg (11 lb). The Promethean P500H by Ibanez is also capable of delivering 500 W into a 4 Ohm load, and weights only 2.9 kg.

Chapter 14

Guitar Amplifier

Mesa Boogie Mark IV, a guitar combo amplifier

A **guitar amplifier** (or **guitar amp**) is an electronic amplifier designed to make the signal of an electric or acoustic guitar louder so that it will produce sound through a loudspeaker. Guitar amplifiers also modify the instrument's tone by emphasizing or de-emphasizing certain frequencies and adding electronic effects.

Amplifiers consist of one or more circuit stages which have unique responsibilities in the modification of the input signal. The power amplifier or output stage produces a high current signal to drive a speaker to produce sound. One or more preamplifier stages precede the power amplifier stage. The preamplifier is a voltage amplifier that amplifies the guitar signal to a level that can drive the power stage. There may be one or more tone stages which affect the character of the guitar signal: before the preamp stage (as in the case of guitar pedals), in between the preamp and power stages (as in the cases of effects loop or many dedicated amplifier tone circuits), in between multiple stacked preamp stages, or in feedback loops from a post-preamp signal to an earlier pre-preamp signal (as in the case of presence modifier circuits). The tone stages may also have electronic effects such as equalization, compression, distortion, chorus, or reverb. Amplifiers may use vacuum tubes (in Britain they are called valves), or solid state (transistor) devices, or both.

There are two configurations of guitar amplifiers: combination ("combo") amplifiers, which include an amplifier and one, two, or four speakers in a wooden cabinet; and the standalone amplifier (often called a "head" or "amp head"), which does not include a speaker, but rather passes the signal to a speaker cabinet or "cab". Guitar amplifiers range in price and quality from small, low-powered practice amplifiers, designed for students, which sell for less than US$50, to expensive "boutique" amplifiers which are custom-made for professional musicians and can cost thousands of dollars.

History

The first electric instrument amplifiers were not designed for use with electric guitars. The earliest examples appeared in the early 1930s when the introduction of electrolytic capacitors and rectifier tubes allowed the production of economical built-in power supplies that could be plugged into wall sockets, instead of heavy multiple battery packs. While guitar amplifiers from the beginning were used to amplify acoustic guitar, electronic amplification of guitar was first widely popularized by the 1930s and 1940s craze for Hawaiian music, which extensively employed the amplified lap steel Hawaiian guitar.

Tone controls on early guitar amplifiers were very simple and provided a great deal of treble boost. but the limited controls, the loudspeakers used, and the low power of the amplifiers (typically 15 watts or less prior to the mid-1950s) gave poor high treble and bass output. Some models also provided effects such as spring reverb and/or an electronic tremolo unit. Early Fender amps labeled tremolo as "vibrato" and labeled the vibrato arm of the Stratocaster guitar as a "tremolo bar".

In the 1960s, guitarists experimented with distortion produced by deliberately overdriving their amplifiers. The Kinks guitarist Dave Davies produced early distortion effects by connecting the already distorted output of one amplifier into the input of another. Later, most guitar amps were provided with preamplifier distortion controls, and "fuzz boxes" and other effects units were engineered to safely and reliably produce these

sounds. In the 2000s overdrive and distortion has become an integral part of many styles of electric guitar playing, ranging from blues rock to heavy metal and hardcore punk.

Guitar amplifiers were at first used with bass guitars and electronic keyboards, but other instruments produce a wider frequency range and need a suitable amplifier and full-range speaker system. Much more amplifier power is required to reproduce low-frequency sound, especially at high volume. Reproducing low frequencies also requires a suitable woofer or subwoofer speaker and enclosure. Woofer enclosures need to be larger and more sturdily built than cabinets for mid-range or high-frequency (tweeter) speakers.

Types

Two combo amplifiers

Guitar amplifiers are manufactured in two main forms. The "combination" (or "combo") amplifier contains the amplifier head and guitar speakers in a single unit which is typically housed in a rectangular wooden box. The amplifier head or "amp head" contains the electronic circuitry constituting the preamp, built-in effects processing, and the power amplifier. Combo amps have at least one 1/4" input jack where the patch cord from the electric guitar can be plugged in. Other jacks may also be provided, such as an additional input jack, "send" and "return" jacks to create an effects loop (for connecting electronic effects such as compression, reverb, etc.), an extension speaker jack (for connecting an additional speaker cabinet). Some smaller practice amps have stereo RCA jacks for connecting a CD player or other sound source and a 1/4" headphone jack so that the player can practice without disturbing neighbours or family members.

Kustom 200 bass amp - amp head and speakers, 100 watts RMS, two channels, two 15" speakers, 1971

Some amplifiers have a line out jack for connecting the amplifier's signal to a PA system or recording console or to connect the amplifier to another guitar amp. In but most styles of rock and blues guitar, the line out is not used to connect the guitar amp to a PA system or recording console, because the tonal coloration and overdrive from the amplifier and speaker is considered an important part of the amplifier's sound. However, players do use the line out to connect one guitar amplifier to another amplifier, in order to create different tone colors or sound effects.

In the "amp head" form, the amplifier head is separate from the speakers, and joined to them by speaker cables. The separate amplifier is called an amplifier head, and is

commonly placed on top of one or more loudspeaker enclosures. A separate amplifier head placed atop a guitar speaker enclosure or guitar speaker cabinet forms an amplifier "stack" or "amp stack". Amp heads may also have the different types of input and output jacks listed above in the combo section. In addition to a 1/4" input jack, acoustic guitar amplifiers typically have an additional input jack for a microphone, which is easily identified because it will use a three-pin XLR connector. Phantom power is not often provided on general-use amps, restricting the choice of microphones for use with these inputs. However, for high-end acoustic amplifiers, phantom power is often provided, so that musicians can use condenser microphones.

Amplifiers used with electric guitars may be solid state, which are lighter in weight and less expensive than tube amplifiers. Many guitarists, particularly in the genres of blues and rock, prefer the sound of vacuum tube amplifiers despite their higher cost, heavier weight, the need to periodically replace tubes and need to re-bias the output tubes (every year or two with moderate use). Some companies design amplifiers that require no biasing as long as properly rated tubes are used. Some modern amplifiers use a mixture of tube and solid-state technologies.

Since the advent of microprocessors and digital signal processing, "modeling amps" have been developed in the late 1990s, these can simulate the sounds of a variety of well-known tube amplifiers without needing to use vacuum tubes. Amplifiers with processors and software emulate the sound of a classic amp well, but from the player's point of view the response of these amplifiers may not feel the same as the digital modeling does not accurately model all aspects of a tube amplifier.

A wide range of instrument amplifiers is available, some for general purposes and others designed for specific instruments or particular sounds. These include:

- **"Traditional" guitar amplifiers**, with a clean, warm sound, a sharp treble roll-off at 5 kHz or less and bass roll-off at 60–100 Hz, and often built-in reverb and tremolo ("vibrato") units. These amplifiers, such as the Fender "Tweed"-style amps, are often used by traditional rock, blues, and country musicians.Traditional amps have more recently become popular with musicians in indie and alternative bands
- **Hard rock-style guitar amplifiers**, which often include preamplification controls, tone filters, and distortion effects that provide the amplifier's characteristic tone. Users of these amplifiers use the amplifier's tone to add "drive", intensity, and "edge" to their guitar sound. Amplifiers of this type, such as Marshall amplifiers, are used in a range of genres, including hard rock, metal, and punk.
- **Bass amplifiers**, with extended bass response and tone controls optimized for bass guitars (or more rarely, for upright bass). Higher-end bass amplifiers sometimes include compressor or limiter features, which help to keep the amplifier from distorting at high volume levels, and an XLR DI output for patching the bass signal directly into a mixing board. Bass amplifiers are often provided with external metal heat sinks or fans to help keep the amplifier cool.

- ***Acoustic*** **amplifiers**, similar in many ways to keyboard amplifiers but designed specifically to produce a "clean," transparent, "acoustic" sound when used with acoustic instruments with built-in transducer pickups and/or microphones.

Vacuum tube amplifiers

The glow from four "Electro Harmonix KT88" brand power tubes lights up the inside of a Traynor YBA-200 guitar amplifier

Vacuum tubes (valves) were by far the dominant active electronic components in most instrument amplifier applications until the 1970s, when semiconductors (transistors) started taking over for performance and economic reasons, including heat and weight reduction, and improved reliability. High-end tube instrument amplifiers have survived as one of few exceptions, because of the sound quality. Typically, one or more dual triodes are used in the preamplifier section in order to provide sufficient voltage gain to offset losses by tone controls and to drive the power amplifier section.

Rear view of a tube (valve) combo guitar amplifier. Visible are two glass output tubes, six smaller preamp tubes in their metal tube retainers, and both the power transformer and the output transformer.

The output tubes are often arranged in a class AB push-pull connection to improve efficiency; this requires another triode or dual triode to split the phase of the signal. The tubes of the power amplifier stage are almost always of the pentode or beam tetrode type (also known as "kinkless tetrodes", hence the KTxx nomenclature). Some high power models use paralleled pairs of output tubes (four or more in total) in push-pull. Except for the light negative feedback from the secondary end of the output transformer to the driver stage, most amplifying stages work in "raw" open-loop mode. Some designs employ current feedback via unbypassed cathode resistors.

Since most tubes show "soft clipping" gain non-linearity, applying an input signal high enough to overdrive any stage tends to produce favorably natural distortion. Today, most vacuum tube amplifiers are based on the ECC83/12AX7/7025 (dual triode) tubes for the preamplifier and driver sections and the EL84/6BQ5 or EL34/6CA7/KT77 or 6L6/KT66 or 6V6 tubes for the power output section. Some use the KT88/6550 beam power tubes in the output stage. The differing codes for equivalent tubes generally reflect those used by

the original European or U.S.A. based manufacturers. These tubes are now mainly manufactured in Russia, China and Eastern European countries. Some amplifiers, such as the Marshall Silver Jubilee, use solid state components in the preamp, most commonly diodes, to create distortion, a design feature known as diode clipping.

Tube instrument amplifiers are often equipped with lower-grade transformers and simpler power regulation circuits than those of hi-fi amplifiers. They are usually not only for cost-saving reasons, but also are considered as a factor in sound creation. For example, a simple power regulation circuit's output tends to sag when there is a heavy load (that is, high output power) and vacuum tubes usually lose gain factors with lower power voltages. This results in a somewhat compressed sound which could be criticized as a "poor dynamic range" in case of hi-fi amplifiers, but could be desirable as "long sustain" of sounds on a guitar amplifier. Some tube guitar amplifiers use a rectifier tube instead of solid-state diodes specifically for this reason.

Some models have a "spring reverb" unit that simulates the reverberation of an echoic ambient. A reverb unit usually consists of one or more coil springs driven by the preamplifier section using a transducer driver similar to a loudspeaker at one end and an electro-magnetic pickup and preamplifier stage at the other end that picks up the long sustaining spring vibration, which is then mixed with the original signal. Some guitar amplifiers have a tremolo control. An internal oscillator generates a low frequency continuous signal which can modulate the input signal's amplitude or the output tubes' bias, thereby producing a tremolo effect.

Tube amps have the following disadvantages in comparison to solid-state amps. They are bulky and heavy, primarily due to the iron in power and output transformers. Solid-state amplifiers still require power transformers, but are usually direct-coupled and don't need output transformers. Glass tubes are fragile, and require more care and consideration when equipment is moved repeatedly. Tube performance can deteriorate slightly over time before eventual catastrophic failure.

When tube vacuum is maintained at a high level, though, excellent performance and life is possible. They are prone to pick up mechanical noises (microphonic noise), although such electro-mechanical feedback from the loudspeaker to the tubes in combo amplifiers may contribute to sound creation. Tubes benefit from a heater warm-up period before the application of high tension anode voltages; this allows the tube cathodes to operate without damage and so prolongs tube life. This is of particular importance for amplifiers with solid state rectifiers.

Tube amps have the following advantages over solid-state amps. Compared to semiconductors, tubes have a very low "drift" (of specs) over a wide range of operating conditions, specifically high heat/high power. Semiconductors are very heat-sensitive by comparison and this fact usually leads to compromises in solid-state amplifier designs. When a tube fails, it is replaceable. While solid state devices are also replaceable, it is usually a much more involved process (i.e., having the amplifier tested by a professional, removing the faulty component, and replacing it).

For working musicians this is usually a huge problem by comparison to looking in the back of a tube amp at the tubes and simply replacing the faulty tube. In addition, tubes can easily be removed and tested, while transistors cannot. Tube amplifiers respond differently from transistor amplifiers when signal levels approach and reach the point of clipping. In a tube-powered amplifier, the transition from linear amplification to limiting is less abrupt than in a solid state unit, resulting in a less grating form of distortion at the onset of clipping. For this reason, some guitarists prefer the sound of an all-tube amplifier; the aesthetic properties of tube versus solid state amps, though, are a topic of debate in the guitarist community.

Solid-state amplifiers

Most inexpensive guitar amplifiers currently produced are based on semiconductor (solid state) circuits, and some designs incorporate tubes in the preamp stage for their subjectively warmer overdrive sound. Tubes create warm overdrive sounds because instead of cutting the peaked signal off, they more or less pull the peaked audio information back (like natural compression) which creates a fuzzy overdrive sound. While this is a desirable attribute in many cases, the tube's characteristic will "color" all the sounds at any volume, unlike solid state. However, solid state in general have the quickest response time, perhaps even more so than modeling amps.

High-end solid state amplifiers are less common, since many professional guitarists tend to favor vacuum tubes. Some jazz guitarists, however, tend to favor the "colder" sound of solid-state amplifiers, preferring not to color the sound of their guitar with the tube distortion and compression so popular with rock, blues, and metal musicians.. Solid-state amplifiers vary in output power, functionality, size, price, and sound quality in a wide range, from practice amplifiers to professional models. Some inexpensive amplifiers have only a single volume control and a one or two tone controls.

Hybrid amplifiers

A tube power amp may be fed by a solid-state pre-amp circuit, as in the Fender Super Champ XD and the Roland Bolt amplifier, which is thereby classed as a 'hybrid' amp. Randall Amplifier's current flagship models, the V2 and T2, use hybrid amp technology. Alternatively, a tube pre-amp can feed a solid state output stage, as in models from Kustom and Vox. This approach dispenses with the need for an output transformer and allow modern power levels to be easily achieved.

The Roland Micro Cube, left, a small and portable digital modeling amplifier

Modeling amplifiers

Modeling amplifiers simulate the sound of well-known guitar amps, cabinets, and effects, as well as simulating the way traditional speaker cabinets sound when mixed with different types of microphones. They may also be an original creation not meant to simulate any particular real world guitar amp at all, instead allowing the user to create their own unique sound. Such as the original creations of companies like AcmeBarGig, or Peavey. This is usually achieved through digital processing, although there are analog modeling amps as well, such as the Tech 21 Trademark. Modeling technology offers several advantages over traditional amplification. A modeling amp typically is capable of a wide range of tones and effects, and offers cabinet simulation, so it can be recorded without a microphone. Most modeling amps digitize the input signal and use a DSP, a dedicated microprocessor, to process the signal with digital computation. Some modeling amps incorporate vacuum tubes, digital processing, and some form of power attenuation.

Acoustic guitar amplifiers

These amplifiers are designed to be used with acoustic guitars, especially for the way these instruments are used in relatively quiet genres such as folk and bluegrass. They are similar in many ways to keyboard amplifiers, in that they have a relatively flat frequency response, and they are usually designed so that neither the power amplifier nor the speakers will introduce additional coloration.

To produce this relatively "clean" sound, these amplifiers often have very powerful amplifiers (providing up to 800 watts RMS), to provide additional "headroom" and prevent unwanted distortion. Since an 800 watt amplifier built with standard Class AB technology would be very heavy, some acoustic amplifier manufacturers use lightweight Class D amplifiers, which are also called "switching amplifiers."

Acoustic amplifiers are designed to produce a "clean", transparent, "acoustic" sound when used with acoustic instruments with built-in transducer pickups and/or microphones. The amplifiers often come with a simple mixer, so that the signals from a pickup and microphone can be blended. Since the early 2000s, it has become increasingly common for acoustic amplifiers to be provided with a range of digital effects, such as reverb and compression. As well, these amplifiers often contain feedback-suppressing devices, such as notch filters or parametric equalizers.

Amplifier configuration

In the case of electric guitars, an amplifier stack consisting of a head atop one cabinet is commonly called a *half stack*, while a head atop two cabinets is referred to as a *full stack*. The cabinet which the head sits on often has an angled top in front, while the lower cabinet of a full stack has a straight front. The first version of the *Marshall stack* was an amp head on an 8x12 cabinet, meaning a single speaker cabinet containing eight 12" guitar speakers. After six of these cabinets were made, the cabinet arrangement was changed to an amp head on two 4x12 cabinets, meaning four 12" speakers, to enable transporting the amp rig.

In heavy metal bands, the term "double stack" or "full stack" is sometimes used to refer to two stacks, with a second amplifier head serving as a slave to the first and four speaker cabinets in total. Another name for the "Head & Cab" that comes from the 1960s and 1970s is "Piggyback". Vox amp stacks could be put on a tiltable frame with casters. Fender heads could be attached to the cab and had "Tilt-Back" legs, like those used on larger Fender combo amps. Typically, a guitar amp's preamplifier section provides sufficient gain so that an instrument can be connected directly to its input, and sufficient power to connect loudspeakers directly to its output, both without requiring extra amplification.

Another arrangement, often used for public address amplifier systems, is to provide two stages of amplification in separate units. First a *preamplifier* or *mixer* is used to boost the instrument output, normally to line level, and perhaps to mix signals from several instruments. The output from this preamplifier is then connected to the input of a power amplifier, which powers the loudspeakers.

Performing musicians that use the "two-stage" approach (as opposed to an amplifier with an integrated preamplifier and power amplifier) often want to custom-design a combination of equipment that best suits their musical or technical needs, and gives them more tonal and technical options. Some musicians require preamps that include specific features. Acoustic performers sometimes require preamps with "notch" filters (to prevent

feedback), reverb, an XLR DI output, or parametric equalization. Hard rock, metal, or punk performers may desire a preamplifier with a range of distortion effects. As well, some musicians have specific power amplifier requirements, such as low-noise design, very high wattage, the inclusion of limiter features to prevent distortion and speaker damage, or biamp-capable operation.

With the "two-stage" approach, the preamplifier and power amplifier are often mounted together in a rack case. This case may be either free-standing or placed on top of a loudspeaker cabinet. If many rack-mounted effects are used, the rack may be a large unit on wheels. Some touring players need several racks of effects units to reproduce on stage the sounds they have produced in the studio. At the other extreme, if a small rack case containing both preamp and power amp is placed on top of a guitar speaker cabinet, the distinction between a rack and a traditional amp head begins to blur. Another variation is to combine the power amplifier into the speaker cabinet, an arrangement called a *powered speaker*, and use these with a separate preamp, sometimes combined into an effects pedal board or floor preamp/processor.

Preamplifiers are also used to connect very low-output or high-impedance instruments to instrument amplifiers. When piezoelectric transducers are used on upright bass or other acoustic instruments, the signal coming directly from the transducer is often too weak and it does not have the correct impedance for direct connection to an instrument amplifier. Small, battery-powered preamps are often used with acoustic instruments to resolve these problems.

Distortion, power, and volume

Power output

For electric guitar amplifiers, there is often a distinction between "practice" or "recording studio" guitar amps, which tend to have output power ratings of 20 watts down to a small fraction of a watt, and "performance" amps, which are generally 50 watts or higher. Traditionally, these have been fixed-power amplifiers, with a few models having a half-power switch to slightly reduce the listening volume while preserving power-tube distortion. The relationship between perceived volume and power output is not immediately obvious. A 5-watt amplifier is perceived to be half as loud as a 50-watt amplifier (a tenfold increase in power), and a half-watt amplifier is a quarter as loud as a 50-watt amp. Doubling the power of an amplifier results in a "just noticeable" increase in volume, so a 100-watt amplifier is held to be only just noticeably louder than a 50-watt amplifier. Such generalizations are also subject to the human ear's tendency to behave as a natural compressor at high volumes.

Power attenuation can be used with either low-power or high-power amplifiers, resulting in variable-power amplifiers. A high-power amplifier with power attenuation can produce power-tube distortion through a wide range of listening volumes. Speaker efficiency is also a major factor affecting a tube amplifier's maximum volume. For bass instruments, higher-power amplifiers are needed to reproduce low-frequency sounds. While an electric

guitarist would be able to play at a small club with a 50-watt amplifier, a bass player performing in the same venue would probably need an amplifier with 200 or more watts.

Distortion and volume

Distortion is a feature available on many guitar amplifiers that is not typically found on keyboard or bass guitar amplifiers. Tube guitar amplifiers can produce distortion through pre-distortion equalization, preamp tube distortion, post-distortion EQ, power-tube distortion, tube rectifier compression, output transformer distortion, guitar speaker distortion, and guitar speaker and cabinet frequency response. Distortion sound or "texture" from guitar amplifiers is further shaped or processed through the frequency response and distortion factors in the microphones (their response, placement, and multi-microphone comb filtering effects), microphone preamps, mixer channel equalization, and compression. Additionally, the basic sound produced by the guitar amplifier can be changed and shaped by adding distortion and/or equalization effect pedals before the amp's input jack, in the effects loop just before the tube power amp, or after the power tubes.

Power-tube distortion

Power-tube distortion is required for amp sounds in some genres. In a standard master-volume guitar amp, as the amp's final or master volume is increased beyond the full power of the amplifier, power tube distortion is produced. The "power soak" approach places the attenuation between the power tubes and the guitar speaker. In the re-amped or "dummy load" approach, the tube power amp drives a mostly resistive dummy load while an additional low power amp drives the guitar speaker. In the isolation box approach, the guitar amplifier is used with a guitar speaker in a separate cabinet. A soundproofed isolation cabinet, isolation box, isolation booth, or isolation room can be used.

Volume controls

A variety of labels are used for level attenuation potentiometers in a guitar amplifier and other guitar equipment. Electric guitars and basses have a volume control to attenuate whichever pickup is selected. There may be two volume controls in parallel to mix the signal levels from the neck and bridge pickups. Rolling back the guitar's volume control also changes the pickup's equalization or frequency response, which can provide pre-distortion equalization.

The simplest guitar amplifiers have only a volume control. Most have at least a gain control and a master volume control. The gain control is equivalent to the distortion control on a distortion pedal, and similarly may have a side-effect of changing the proportion of bass and treble sent to the next stage.

A simple amplifier's tone controls typically include passive bass and treble controls. In some cases, a midrange control is provided. The amplifier's master volume control restricts the amount of signal permitted through to the driver stage and the power

amplifier. When using a power attenuator with a tube amplifier, the master volume no longer acts as the master volume control. Instead, the power attenuator's attenuation control controls the power delivered to the speaker, and the amplifier's master volume control determines the amount of power-tube distortion. Power-supply based power reduction is controlled by a knob on the tube power amp, variously labeled "Wattage", "Power", "Scale", "Power Scale", or "Power Dampening".

Use with other instruments

A 3 x 6 stack of Marshall guitar cabinets for Jeff Hanneman of Slayer. Such large stacks are often for impressive appearance rather than sound, as most of the cabinets will not be connected.

Musicians often run sound-sources other than guitars through guitar amps. For live performances, synthesizers and drum machines or keyboards are often put through guitar amps to create a richer sound than can be obtained by patching the direct-outs right into the PA system. Guitar amplifiers can add tonal coloration, roll off unwanted high frequencies, and add overdrive or distortion. Deep Purple's Jon Lord played his Hammond Organ through a distorted Marshall amp to create a sound more suitable for heavy rock. String instruments and vocals are also put through guitar amps to add distortion effects. Some blues harp players also use guitar amps to create a warmer overdrive sound for their harmonica playing; 1950s-style "tweed" amps are often used for this purpose, such as Fender Bassman combo amps.

Recording engineers occasionally run pre-recorded parts through miked guitar amps, a process called re-amping. When a guitar part is recorded "dry" (e.g., without effects or

distortion), straight into the mixing board for a recording, this gives the producer and mixing engineer much more flexibility to create new re-mixes or new tones from the recording. If a guitar player records an electric guitar part that is run through a chorus pedal and a distortion pedal, there is little that can be done at the "mix-down" stage to change the sound of this recording, beyond "tweaking" the equalization and modifying the level. Since re-mixing or mixdown can take place weeks, months, or even years after the original recording session, it may be impossible to have the guitarist come in to re-record a new part.

If the dry guitar sound is recorded, though, the mixing engineers can add any effects they want to the signal and then re-play it through a miked guitar amplifier which is being recorded. The effects, amplifiers, cabinets, and miking processes can be changed to any combination. When a dry guitar has been recorded, it can be a useful tool for "updating" an older recording. For example, if a band wants to re-release a 1980s-era album on which the guitar has a very dated 1980s sound, with heavy flanging and artificial-sounding electronic distortion, the band can update the guitar sound by re-amping the dry signal and using 2000s-era effects.

Mixing guitar amp signals with other signals is also done by some musicians. Chris Squire of Yes produced his bass guitar sound by playing through a guitar amplifier with its bass turned down, treble turned up, and volume turned up well into distortion; the miked guitar speaker signal was then mixed with a direct-input (DI) signal, a technique that has been used for processing synth keyboards as well. A bass guitar can also be played through a bass amp in parallel with a distorted guitar amp by using a DI box; the bass amp provides the low frequencies, while the guitar amp – which is not capable of reproducing the lowest frequencies of the bass guitar– emphasizes the upper harmonics of the instrument's tone.

Chapter 15

Instrument Amplifier

An **instrument amplifier** is an electronic amplifier that converts the often barely audible or purely electronic signal from musical instruments such as an electric guitar, an electric bass, or an electric keyboard into an electronic signal capable of driving a loudspeaker that can be heard by the performers and audience. Combination ("combo") amplifiers include a preamplifier, a power amplifier, tone controls, and one or more speakers in a cabinet, a housing usually made of hardwood, plywood, particleboard, or, less commonly, moulded plastic. Instrument amplifiers for some instruments are also available without an internal speaker; these amplifiers have to be plugged into an external speaker cabinet.

A Fender "combo" amplifier being picked up with a microphone in a recording studio

Instrument amplifiers are available for specific instruments, including the electric guitar, electric bass, electric keyboards, and acoustic instruments such as the mandolin and banjo. Some amplifiers are designed for specific styles of music, such as the "traditional"-style "tweed" guitar amplifiers used by blues and country musicians, and the Marshall amplifiers used by hard rock and heavy metal bands.

Unlike home "hi-fi" amplifiers or public address systems, which are designed to reproduce accurately the source sound signals with as little harmonic distortion as possible, instrument amplifiers are often designed to add additional tonal coloration to the original signal or emphasize (or de-emphasize) certain frequencies. The two exceptions are keyboard amplifiers and "acoustic" instrument amplifiers, which typically aim for a relatively flat frequency response.

Types

Guitar amplifiers

Standard amps

A small Gibson "combo" amplifier

Standard amplifiers, such as the Fender "tweed"-style amps and Gibson amps, are often used by traditional rock, blues, and country musicians who wish to create a "vintage" 1950s-style sound. They are used by electric guitarists, pedal steel guitar players, and blues harmonica ("harp") players. Combo amplifiers such as the Fender Super Reverb have tube amplifiers, four 10" speakers, and built-in reverb and "vibrato" effects units.

These amps are designed to produce a variety of sounds ranging from a clean, warm sound (when used in country and soft rock) to a growling, natural overdrive, when the

volume is set near its maximum, (when used for blues, rockabilly, and roots rock). These amplifiers usually have a sharp treble roll-off at 5 kHz to reduce the extreme high frequencies, and a bass roll-off at 60–100 Hz to reduce boominess. The nickname "tweed" refers to the lacquered beige-light brown fabric covering used on these amplifiers.

The Roland "Jazz Chorus" amplifier

The smallest "combo" amplifiers, which are mainly used for rehearsal and warm-up purposes, may have only a single 8" or 10" speaker. Some harmonica players use these small combo amplifiers for concert performances, though, because it is easier to create natural overdrive with these lower-powered amplifiers. Larger combo amplifiers, with one 12 inch speaker or two or four 10 or 12 inch speakers are used for club performances. For large concert venues, performers may also use an amplifier "head" with several separate speaker cabinets (which usually contain two or four 12" speakers).

Hard rock and heavy metal

These electric guitar amplifiers add an aggressive "drive", intensity, and "edge" to the guitar sound with distortion effects, preamplification boost controls, and tone filters. While many of the most expensive, high-end models use tube amplifiers, there also many models that use transistor amplifiers, or a mixture of the two technologies (i.e., a tube preamplifier with a transistor power amplifier).

A 3 x 6 "stack" of Marshall guitar cabinets on stage

Amplifiers of this type, such as Marshall amplifiers, are used in a range of the louder, heavier genres of rock, including hard rock, heavy metal, and hardcore punk. This type of amplifier is available in a range of formats, ranging from smaller combo amplifiers for rehearsal and warm-up purposes to heavy "heads" which are designed to be used with separate speaker cabinets, which is colloquially referred to as a "stack." In the late 1960s and early 1970s, public address systems at rock concerts were used mainly for the vocals. As a result, to get a loud electric guitar sound, early heavy metal and rock-blues bands often used "stacks" of 4x12" Marshall speaker cabinets on the stage. In 1969 Jimi Hendrix used four stacks to create a powerful lead sound, and in the early 1970s by the band Blue Öyster Cult used an entire wall of Marshall Amplifiers to create a roaring wall of sound. In the 1980s, metal bands such as Slayer and Yngwie Malmsteen also used "walls" of over 20 Marshall cabinets. However, by the 1980s and 1990s, most of the sound at live concerts was produced by the sound reinforcement system rather than the onstage guitar amplifiers, so most of these cabinets were not connected to an amplifier. Instead, walls of speaker cabinets were used for aesthetic reasons.

A Mesa-Boogie "Mark IV" amplifier

Amplifiers for harder, heavier genres often use valve amplifiers (known as "tube amplifiers" in North America) also. Valve amplifiers have a "warmer" tone than those of transistor amps, particularly when overdriven. Instead of abruptly clipping off the signal at cut-off and saturation levels, the signal is rounded off more smoothly. Vacuum tubes also exhibit different harmonic effects than transistors. In contrast to the "tweed"-style amplifiers, which use speakers in an open-backed cabinet, companies such as Marshall tend to use 12" speakers in a closed-back cabinet. These amplifiers usually allow users to switch between "clean" and distorted tones (or a rhythm guitar-style "crunch" tone and a sustained "lead" tone) with a foot-operated switch.

An "Orange" brand speaker cabinet and separate amplifier "head" with its distinctive orange Tolex covering.

Bass

Bass amplifiers are designed for bass guitars or more rarely, for upright bass. They differ from amplifiers for the regular electric guitar in several respects. They have extended bass response and tone controls optimised for bass instruments, which produce pitches of 40 Hz, in the case of a standard four-string electric bass, or even lower for five- or six-string electric basses.

Higher-end bass amplifiers sometimes include compressor or limiter features, which help to keep the amplifier from distorting at high volume levels, and an XLR DI output for patching the bass signal directly into a mixing board or PA systems. Larger, more powerful bass amplifiers (300 or more watts) are often provided with external metal heat sinks or fans to help keep the amplifier cool.

A 2 x 10" bass speaker cabinet stacked on top of a 15" cabinet, with separate bass amplifier "head" unit

Speaker cabinets designed for bass instrument amplification usually use larger loudspeakers (or more loudspeakers, in the case of the popular 4 X 10" cabinets, which contain four 10" speakers) than the cabinets used for other instruments, so that they can move the larger amounts of air needed to reproduce low frequencies. While the largest speakers commonly used for regular electric guitar are 12" speakers, electric bass speaker cabinets often use 15" speakers. Bass players who play styles of music that require an extended low-range response, such as death metal, sometimes use speaker cabinets with 18" speakers.

The speakers used for bass instrument amplification tend to be more heavy-duty than speakers used for regular electric guitar, and the speaker cabinets are typically more rigidly constructed and heavily braced, to prevent unwanted buzzes and rattles. Bass cabinets often include bass reflex ports or openings in the cabinet, which improve the bass response, especially at high volumes.

Keyboard

This type of amplifier is used to amplify a range of electric and electronic keyboards, such as synthesizers, Hammond organ-style keyboards, stage pianos and electric pianos. Since keyboard instruments contain a wide frequency range, from very low bass notes to extremely high pitches, keyboard amplifiers are often provided with a large woofer speaker to handle the low notes and a horn (or tweeter) for the high notes.

Keyboard amplifiers intended for general use for a range of keyboard applications usually have very low distortion and extended, flat frequency response in both directions. The exception to this rule is keyboard amplifiers designed for the Hammond organ, such as the vintage Leslie speaker cabinet and modern recreations, which have a tube amplifier which is often turned up to add a warm, "growling" overdrive to the organ sound.

Unlike bass amplifiers and electric guitar amplifiers, keyboard amplifers are rarely used in the "amplifier head" and separate speaker cabinets configuration. Instead, most keyboard amplifiers are "combo" amplifiers that integrate the amplifier, tone controls, and speaker into a single wooden cabinet. Another unusual aspect of keyboard amplifiers is that they are often designed with a "wedge" shape, as used with monitor speakers. This allows the cabinet to be rocked back so that it will project sound upwards at a roughly 45' angle, which is more suitable for a seated keyboardist.

Keyboard amplifiers often have a simple onboard mixer with multiple inputs, so that keyboardists can control the tone and level of several keyboards. In some genres, such as progressive rock, for example, keyboardists may perform with several synthesizers, electric pianos, and electro-mechanical keyboards.

A small amp capable of mixing the inputs from two keyboards. This amplifier would be suitable for at-home practice, but it would not be loud enough for a rehearsal or performance.

Acoustic

These amplifiers are designed to be used with acoustic instruments such as violin ("fiddle"), mandolin, and acoustic guitar, especially for the way these instruments are used in relatively quiet genres such as folk and bluegrass. They are similar in many ways to keyboard amplifiers, in that they have a relatively flat frequency response, and they are usually designed so that neither the power amplifier nor the speakers will introduce additional coloration.

To produce this relatively "clean" sound, these amplifiers often have very powerful amplifiers (providing up to 800 watts RMS), to provide additional "headroom" and prevent unwanted distortion. Since an 800 watt amplifier built with standard Class AB technology would be very heavy, some acoustic amplifier manufacturers use lightweight Class D amplifiers, which are also called "switching amplifiers."

Acoustic amplifiers are designed to produce a "clean", transparent, "acoustic" sound when used with acoustic instruments with built-in transducer pickups and/or microphones. The amplifiers often come with a simple mixer, so that the signals from a pickup and microphone can be blended. Since the early 2000s, it has become increasingly common for acoustic amplifiers to be provided with a range of digital effects, such as reverb and compression. As well, these amplifiers often contain feedback-suppressing devices, such as notch filters or parametric equalizers.

Roles

Instrument amplifiers are designed for a different purpose than 'Hi-Fi' (high fidelity) stereo amplifiers used for radios and home stereo systems. Hi-fi home stereo amplifiers are designed to accurately reproduce the source sound signals from pre-recorded music, with as little harmonic distortion as possible. In contrast, instrument amplifiers are often designed to add additional tonal coloration to the original signal or emphasize certain frequencies. For electric instruments such as electric guitar, the amplifier helps to create the instrument's tone by boosting the input signal gain and distorting the signal, and by emphasizing frequencies deemed to be desirable (e.g., low frequencies) and de-emphasizing frequencies deemed to be undesirable (e.g., very high frequencies).

The two exceptions are keyboard amplifiers and acoustic amplifiers which are used by folk and bluegrass musicians for amplifying acoustic instruments such as acoustic guitar, violin, and mandolin. Acoustic amplifiers typically aim for a relatively flat response across the entire frequency range, much like a Public Address system.

Size and power rating

In the 1960s and 1970s, large, heavy, high output power amplifiers were preferred for instrument amplifiers, especially for large concerts, because public address systems were generally only used to amplify the vocals. Moreover, in the 1960s, PA systems typically did not use monitor speaker systems to amplify the music for the onstage musicians.

Instead, the musicians were expected to have instrument amplifiers that were powerful enough to provide amplification for the stage and audience. In late 1960s and early 1970s rock concerts, bands often used large stacks of speaker cabinets powered by heavy tube amplifiers such as the Super Valve Technology (SVT) amplifier, which was often used with eight 10" speakers.

However, over subsequent decades, PA systems were substantially improved, and different approaches such as horn-loaded "bass bins" (in the 1980s) and subwoofers (1990s and 2000s) were used to amplify bass frequencies. As well, in the 1980s and 1990s, monitor systems were substantially improved, which allowed sound engineers to provide onstage musicians with a loud, clear, and full-range reproduction of their instruments' sound.

As a result of the improvements to PA systems and monitor systems, musicians in the 2000s no longer need to have huge, powerful amplifier systems; a small combo amplifier patched into the PA will suffice. In the 2000s, virtually all of the sound reaching the audience in large venues comes from the PA system. As well, in the 2000s onstage instrument amplifiers are more likely to be kept at a low volume, because high volume levels onstage makes it harder to control the sound mix and produce a clean sound.

As a result, in many large venues much of the onstage sound reaching the musicians now comes from the monitor speakers, not from the instrument amplifiers. While stacks of huge speaker cabinets and amplifiers are still used in concerts (especially in heavy metal), this is often mainly for aesthetics or to create a more authentic tone. The switch to smaller instrument amplifiers makes it easier for musicians to transport their equipment to performances. As well, it makes concert stage management easier at large clubs and festivals where several bands are performing in sequence, because the bands can be moved on and off the stage more quickly.

Amplifier technology

Instrument amplifiers may be based on thermionic ("tube" or "valve") or solid state (transistor) technology.

Tube Amplifiers

Vacuum tubes were the dominant active electronic components in amplifiers manufactured from the 1930s through the early 1970s, and tube amplifiers continue to be preferred by some professional musicians. Some musicians believe that tube amplifiers produce a "warmer" or more "natural" sound than solid state units. However, these subjective assessments of the attributes of tube amplifiers' sound qualities are the subject of ongoing debate.

Although tube amplifiers produce more heat than solid state amplifiers, few manufacturers of these units include cooling fans in the chassis. While tube amplifiers do

need to attain a proper operating temperature, if the temperature goes above this operating temperature, it may shorten the tubes' lifespan and lead to tonal inconsistencies.

A Trace Elliot "Bonneville" tube amplifier as seen from the rear view: note the vacuum tubes extending into the wooden cabinet.

Solid-state Amplifiers

By the 1960s and 1970s, semiconductor transistor-based amplifiers began to become more popular because they are less expensive, lighter-weight, and require less maintenance. In some cases, tube and solid-state technologies are used together in amplifiers. A common setup is the use of a tube preamplifier with a solid-state power amplifier. There are also an increasing range of products that use digital signal processing and digital modeling technology to simulate many different combinations of amp and cabinets.

The output transistors of solid-state amplifiers can be passively cooled by using metal fins called heatsinks to radiate away the heat. For high-wattage amplifiers (over 800 watts), a fan is often used to move air across internal heatsinks.