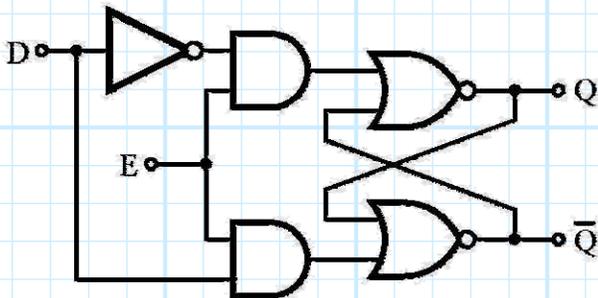


EECS 412 Introduction

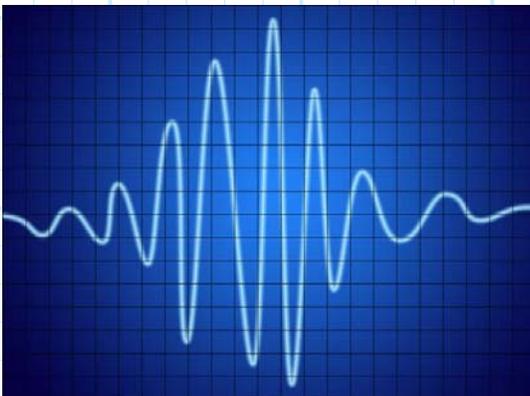
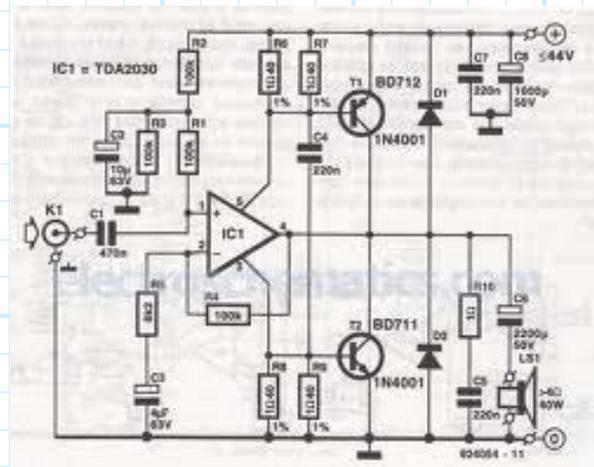
Q: *So what's this class all about? What is its purpose?*

A: In EECS 312 you learned about:



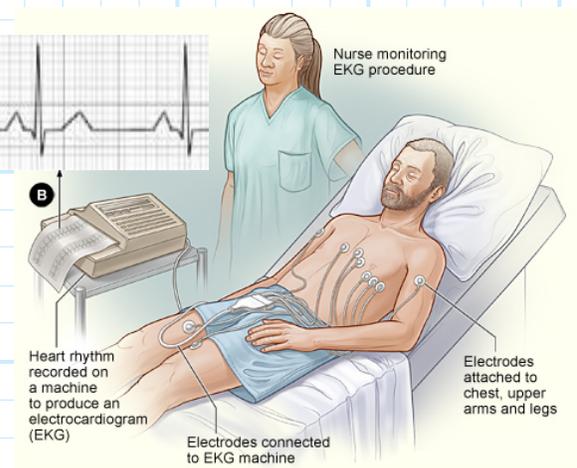
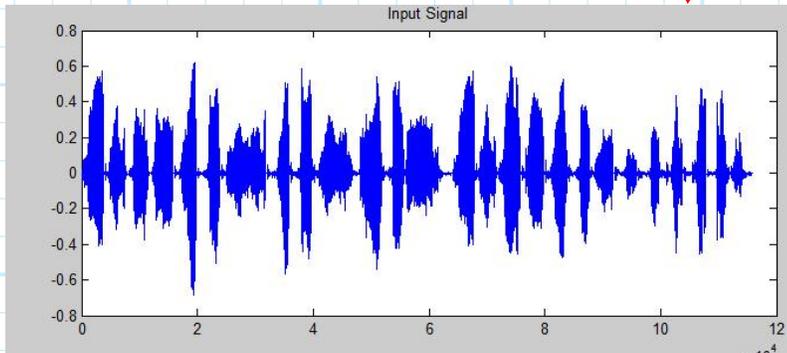
- * Electronic **devices** (e.g., transistors and diodes)
- * How we use transistors to make **digital devices** (e.g., inverters, gates, flip-flops, and memory).

In contrast, EECS 412 will teach you how we use transistors to make **analog devices** (e.g., amplifiers, filters, summers, integrators, etc.).

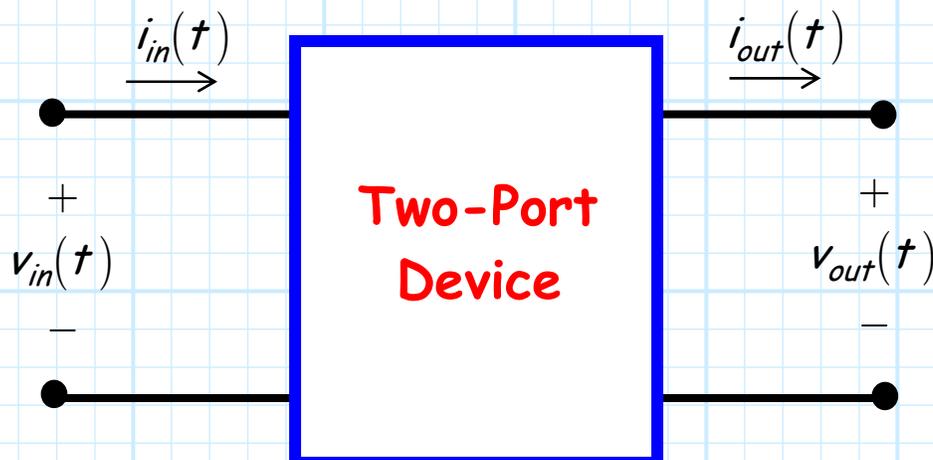


Analog circuits and devices operate on analog signals—usually voltage signals—that represent a **continuous**, time-varying analog of some physical function.

For **example**, the analog voltage signal $v(t)$ can represent an audio pressure wave (i.e., sound), or the beating of a human heart.



Quite often, an analog device has two ports—an **input port** and an **output port**:



A fundamental question in electrical engineering is **determining** the **output** signal $v_{out}(t)$ when the input signal $v_{in}(t)$ is known.

This is frequently a difficult question to answer, but it becomes significantly easier if the two-port device is constructed of **linear**, time-invariant circuit elements!

HO: THE LINEAR, TIME-INVARIANT CIRCUIT

Linear circuit behavior would be not at all useful except for the **unfathomably important** concept of signal expansion via **basis functions**!

HO: SIGNAL EXPANSIONS

Linear systems theory is useful for electrical engineers because most **analog devices and systems are linear** (at least approximately so!).

HO: LINEAR CIRCUIT ELEMENTS

The most powerful tool for analyzing linear systems is its **Eigen function**.

HO: THE EIGEN FUNCTION OF LINEAR SYSTEMS

Complex voltages and currents at times cause much **head scratching**; let's make sure we know what these complex values and functions **physically** mean.

HO: A COMPLEX REPRESENTATION OF SINUSOIDAL FUNCTIONS

Signals may **not** have the explicit form of an Eigen function, **but** our linear systems theory allows us to (relatively) easily analyze this case as well.

HO: ANALYSIS OF CIRCUITS DRIVEN BY ARBITRARY FUNCTIONS

If our linear system is a linear **circuit**, we can apply **basic** circuit analysis to determine all its **Eigen values!**

HO: THE EIGEN SPECTRUM OF LINEAR CIRCUITS

A more general form of the Fourier Transform is the **Laplace** Transform.

HO: THE EIGEN VALUES OF THE LAPLACE TRANSFORM

The numerical value of **frequency** ω has tremendous practical ramifications to us EEs.

HO: FREQUENCY BANDS

A set of **four** Eigen values can completely characterize a two-port linear system.

HO: THE IMPEDANCE AND ADMITTANCE MATRIX

A really important linear (sort of) device is the **amplifier**.

HO: THE AMPLIFIER

The two most important parameter of an amplifier is its **gain** and its **bandwidth**.

HO: AMPLIFIER GAIN AND BANDWIDTH

Amplifier circuits can be quite complex; however, we can use a relatively simple **equivalent circuit** to analyze the result when we connect things to them!

HO: CIRCUIT MODELS FOR AMPLIFIERS

One very useful application of the circuit model is to analyze and characterize types of amplifiers.

HO: CURRENT AND VOLTAGE AMPLIFIERS

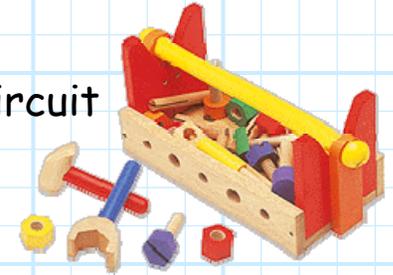
It turns out that amplifiers are only **approximately linear**. It is important that we understand their **non-linear** characteristics and properties.

HO: NON-LINEAR BEHAVIOR OF AMPLIFIERS

Linear Circuits

Many analog devices and circuits are **linear** (or approximately so).

Let's make sure that we understand what this term means, as if a circuit is linear, we can apply a large and helpful **mathematical** toolbox!



Mathematicians often speak of **operators**, which is "mathspeak" for any mathematical operation that can be applied to a single **element** (e.g., value, variable, vector, matrix, or function).

...operators, operators, operators!!

For example, a **function** $f(x)$ describes an operation **on** variable x (i.e., $f(x)$ is operator on x). E.G.:

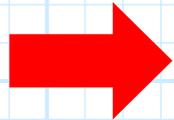
$$f(y) = y^2 - 3 \quad g(t) = 2t \quad y(x) = |x|$$

Functions can be operated on

Moreover, we find that functions can likewise be operated on!

For example, **integration** and **differentiation** are likewise mathematical operations—operators that operate **on functions**. E.G.:

$$\int f(y) dy \quad \frac{d g(t)}{dt} \quad \int_{-\infty}^{\infty} |y(x)| dx$$



A special and very important class of operators are **linear operators**.

Linear operators are **denoted** as $\mathcal{L}[y]$, where:

- * \mathcal{L} symbolically denotes the mathematical **operation**;
- * And y denotes the **element** (e.g., function, variable, vector) being **operated on**.

We call this linear superposition

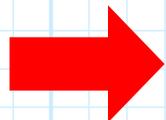
A **linear** operator is any operator that satisfies the following **two** statements for any and **all** y :

1. $\mathcal{L}[y_1 + y_2] = \mathcal{L}[y_1] + \mathcal{L}[y_2]$
2. $\mathcal{L}[a y] = a \mathcal{L}[y]$, where a is any constant.

From these two statements we can **likewise** conclude that a linear operator has the property:

$$\mathcal{L}[a y_1 + b y_2] = a \mathcal{L}[y_1] + b \mathcal{L}[y_2]$$

where both a and b are constants.



Essentially, a linear operator has the property that any weighted sum of solutions is **also** a solution!

An example of a linear function

For **example**, consider the function:

$$\mathcal{L}[t] = g(t) = 2t$$

At $t = 1$:

$$g(t = 1) = 2(1) = 2$$

and at $t = 2$:

$$g(t = 2) = 2(2) = 4$$

Now at $t = 1 + 2 = 3$ we find:

$$\begin{aligned} g(1 + 2) &= 2(3) \\ &= 6 \\ &= 2 + 4 \\ &= g(1) + g(2) \end{aligned}$$

See, it works like it's suppose to!

More generally, we find that:

$$\begin{aligned}g(t_1 + t_2) &= 2(t_1 + t_2) \\ &= 2t_1 + 2t_2 \\ &= g(t_1) + g(t_2)\end{aligned}$$

and

$$\begin{aligned}g(at) &= 2at \\ &= a2t \\ &= ag(t)\end{aligned}$$

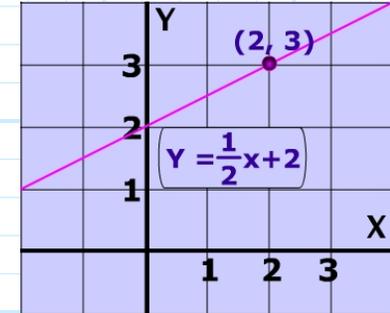
Thus, we conclude that the function $g(t) = 2t$ is **indeed** a **linear** function!

Surely this is linear

Now consider **this** function:

$$y(x) = mx + b$$

Q: *But that's the equation of a line! That must be a linear function, right?*



A: I'm not sure—let's find out!

We find that:

$$\begin{aligned} y(ax) &= m(ax) + b \\ &= amx + b \end{aligned}$$

but:

$$\begin{aligned} ay(x) &= a(mx + b) \\ &= amx + ab \end{aligned}$$

therefore:

$$y(ax) \neq ay(x) !!!$$

It's not; and stop calling me Shirley

Likewise:

$$\begin{aligned}y(x_1 + x_2) &= m(x_1 + x_2) + b \\ &= mx_1 + mx_2 + b\end{aligned}$$

but:

$$\begin{aligned}y(x_1) + y(x_2) &= (mx_1 + b) + (mx_2 + b) \\ &= mx_1 + mx_2 + 2b\end{aligned}$$



therefore:

$$y(x_1 + x_2) \neq y(x_1) + y(x_2) \quad !!!$$

 The equation of a line is **not** a linear function!

Moreover, **you** can show that the functions:

$$f(y) = y^2 - 3 \qquad y(x) = |x|$$

are likewise **non-linear**.

The derivative is a linear operator

Remember, linear operators need **not** be functions.

Consider the derivative operator, which operates on functions.

$$\frac{d f(x)}{dx}$$

Note that:

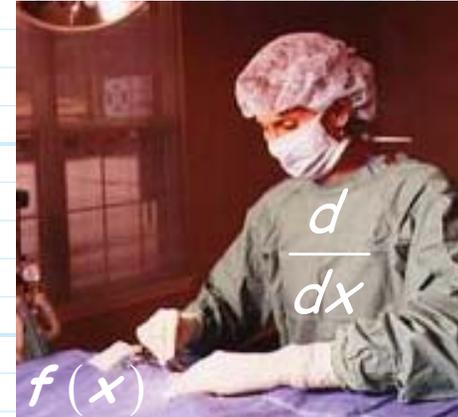
$$\frac{d}{dx}[f(x) + g(x)] = \frac{d f(x)}{dx} + \frac{d g(x)}{dx}$$

and also:

$$\frac{d}{dx}[a f(x)] = a \frac{d f(x)}{dx}$$

We thus can conclude that the **derivative** operation is a **linear operator on function** $f(x)$:

$$\frac{d f(x)}{dx} = \mathcal{L}[f(x)]$$



Most operators are not linear

You can likewise show that the **integration** operation is likewise a **linear operator**:

$$\int f(y) dy = \mathcal{L}[f(y)]$$

But, **you** will find that operations such as:

$$\frac{d g^2(t)}{dt} \quad \int_{-\infty}^{\infty} |y(x)| dx$$

are **not** linear operators (i.e., they are **non-linear** operators).

We find that **most** mathematical operations are in fact **non-linear**!

Linear operators are thus form a small **subset** of all possible mathematical operations.

Linear operators allow for “easy” evaluation

Q: *Yikes! If linear operators are so rare, are we wasting our time learning about them??*

A: Two reasons!

Reason 1: In electrical engineering, the behavior of most of our fundamental **circuit elements** are described by **linear operators**—linear operations are prevalent in **circuit analysis!**

Reason 2: To our great relief, the two characteristics of linear operators allow us to **perform** these mathematical operations with **relative ease!**



Signal Expansions

Q: *How is performing a **linear** operation easier than performing a **non-linear** one??*

A: The "secret" lies in the result:

$$\mathcal{L}[a y_1 + b y_2] = a \mathcal{L}[y_1] + b \mathcal{L}[y_2]$$

Note here that the linear operation performed on a relatively **complex** element $a y_1 + b y_2$ can be determined immediately from the result of operating on the "**simple**" elements y_1 and y_2 .

To see how this might work, let's consider some **arbitrary** function of **time** $v(t)$, a function that exists over some **finite** amount of time T (i.e., $v(t) = 0$ for $t < 0$ and $t > T$).

Say we wish to perform some **linear** operation on this function:

$$\mathcal{L}[v(t)] = ??$$

Complex signals as collections of simple elements



Depending on the **difficulty** of the operation \mathcal{L} , and/or the **complexity** of the function $v(t)$, directly performing this operation could be very **painful** (i.e., approaching impossible).

Instead, we find that we can often **expand** a very complex and **stressful** function in the following way:

$$v(t) = a_0 \psi_0(t) + a_1 \psi_1(t) + a_2 \psi_2(t) + \dots = \sum_{n=-\infty}^{\infty} a_n \psi_n(t)$$

where the values a_n are **constants** (i.e., coefficients), and the functions $\psi_n(t)$ are known as **basis functions**.



Ms. Nomial's first name is Poly

For example, we could **choose** the basis functions:

$$\psi_n(t) = t^n \quad \text{for } n \geq 0$$

Resulting in a **polynomial** of variable t :

$$v(t) = a_0 + a_1 t + a_2 t^2 + a_3 t^3 + \dots = \sum_{n=0}^{\infty} a_n t^n$$

This signal expansion is of course know as the **Taylor Series** expansion.

Choose your basis - but choose wisely



However, there are **many other** useful expansions (i.e., many other useful basis $\psi_n(t)$).

- * The key thing is that the basis functions $\psi_n(t)$ are **independent** of the function $v(t)$. That is to say, the basis functions are **selected** by the engineer doing the analysis (i.e., **you**).
- * The set of selected basis functions form what's known as a **basis**. With this basis we can **analyze** the function $v(t)$.
- * The **result** of this analysis provides the **coefficients** a_n of the signal expansion. Thus, the coefficients **are** directly dependent on the form of function $v(t)$ (as well as the basis used for the analysis). As a result, the set of coefficients $\{a_1, a_2, a_3, \dots\}$ **completely describe** the function $v(t)$!

It's simpler to operate on each element

Q: *I don't see why this "expansion" of function of $v(t)$ is helpful, it just looks like a lot more work to me.*

A: Consider what happens when we wish to perform a **linear** operation on this function:

$$\mathcal{L}[v(t)] = \mathcal{L}\left[\sum_{n=-\infty}^{\infty} a_n \psi_n(t)\right] = \sum_{n=-\infty}^{\infty} a_n \mathcal{L}[\psi_n(t)]$$

Look what happened!

Instead of performing the linear operation on the arbitrary and **difficult** function $v(t)$, we can apply the operation to **each** of the individual basis functions $\psi_n(t)$.

Choose a basis that makes this "easy"

Q: *And that's supposed to be easier??*

A: It **depends** on the linear operation and on the basis functions $\psi_n(t)$.

Hopefully, the operation $\mathcal{L}[\psi_n(t)]$ is **simple** and straightforward.

Ideally, the solution to $\mathcal{L}[\psi_n(t)]$ is **already known!**

Q: *Oh yeah, like I'm going to get so lucky. I'm sure in all my circuit analysis problems evaluating $\mathcal{L}[\psi_n(t)]$ will be long, frustrating, and **painful**.*



A: Remember, **you** get to choose the **basis** over which the function $v(t)$ is analyzed.

A **smart** engineer will **choose** a basis for which the operations $\mathcal{L}[\psi_n(t)]$ are simple and **straightforward!**

This basis is quite popular

Q: *But I'm still confused. How do I choose what basis $\psi_n(t)$ to use, and how do I analyze the function $v(t)$ to determine the coefficients a_n ??*

A: Perhaps an **example** would help. Among the **most popular** basis is this one:

$$\psi_n = \begin{cases} e^{j\left(\frac{2\pi n}{T}\right)t} & 0 \leq t \leq T \\ 0 & t \leq 0, t \geq T \end{cases}$$

and:

$$a_n = \frac{1}{T} \int_0^T v(t) \psi_n^*(t) dt = \frac{1}{T} \int_0^T v(t) e^{-j\left(\frac{2\pi n}{T}\right)t} dt$$

So therefore:

$$v(t) = \sum_{n=-\infty}^{\infty} a_n e^{j\left(\frac{2\pi n}{T}\right)t} \quad \text{for } 0 \leq t \leq T$$



The **astute** among you will recognize this signal expansion as the **Fourier Series!**

It has a very important property!

Q: *Yes, just why is Fourier analysis so prevalent?*

A: The answer reveals itself when we apply a **linear operator** to the signal expansion:

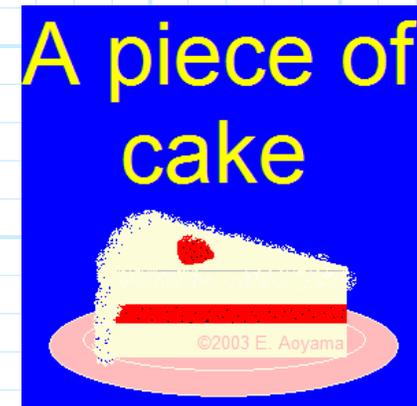
$$\mathcal{L}[v(t)] = \mathcal{L}\left[\sum_{n=-\infty}^{\infty} a_n e^{-j\left(\frac{2\pi n}{T}\right)t}\right] = \sum_{n=-\infty}^{\infty} a_n \mathcal{L}\left[e^{-j\left(\frac{2\pi n}{T}\right)t}\right]$$

Note then that we must **simply** evaluate:

$$\mathcal{L}\left[e^{-j\left(\frac{2\pi n}{T}\right)t}\right]$$

for all n .

We will find that **performing** almost any linear operation \mathcal{L} on basis functions of this type to be exceeding **simple** (more on this later)!



Linear Circuit Elements

Most microwave devices can be described or modeled in terms of the **three** standard circuit elements:

1. RESISTANCE (R) 

2. INDUCTANCE (L) 

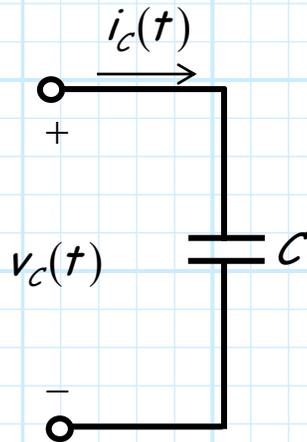
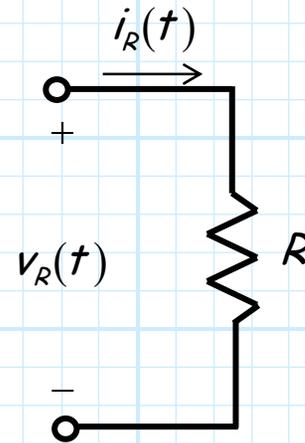
3. CAPACITANCE (C) 

For the purposes of circuit analysis, each of these three elements are **defined** in terms of the **mathematical** relationship between the difference in electric potential $v(t)$ between the two terminals of the device (i.e., the **voltage** across the device), and the **current** $i(t)$ flowing through the device.

We find that for these three circuit elements, the relationship between $v(t)$ and $i(t)$ can be expressed as a linear operator!

$$\mathcal{L}_y^R[v_R(t)] = i_R(t) = \frac{v_R(t)}{R}$$

$$\mathcal{L}_z^R[i_R(t)] = v_R(t) = R i_R(t)$$

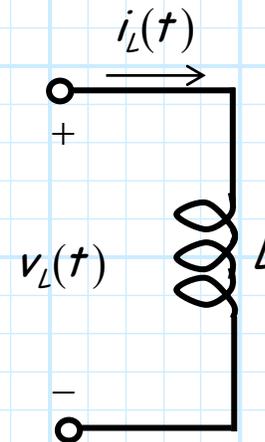


$$\mathcal{L}_y^C[v_C(t)] = i_C(t) = C \frac{dv_C(t)}{dt}$$

$$\mathcal{L}_z^C[i_C(t)] = v_C(t) = \frac{1}{C} \int_{-\infty}^t i_C(t') dt'$$

$$\mathcal{L}_y^L[v_L(t)] = i_L(t) = \frac{1}{L} \int_{-\infty}^t v_L(t') dt'$$

$$\mathcal{L}_z^L[i_L(t)] = v_L(t) = L \frac{di_L(t)}{dt}$$

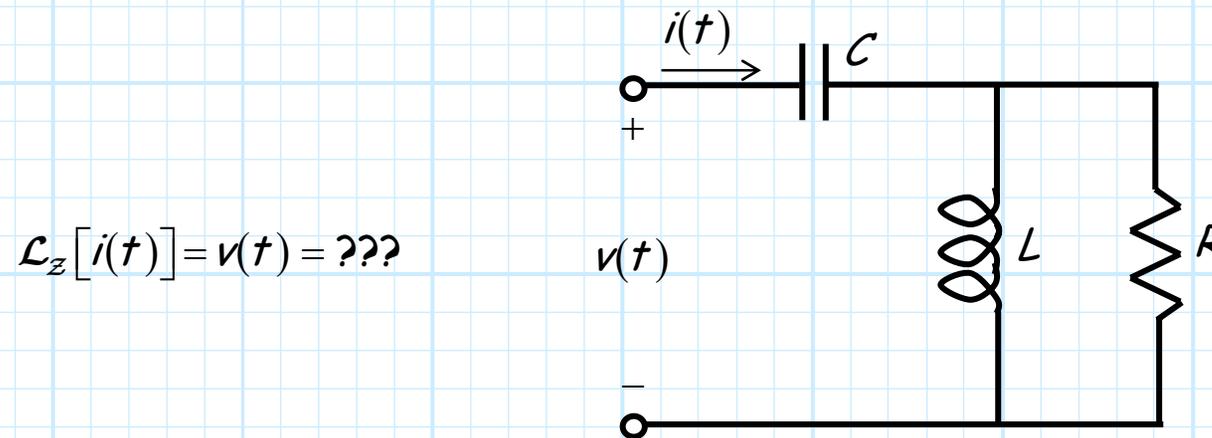


Since the circuit behavior of these devices can be expressed with **linear** operators, these devices are referred to as **linear circuit elements**.

A linear operator describes any relationship

Q: Well, that's simple enough, but what about an element formed from a **composite** of these fundamental elements?

For **example**, for example, how are $v(t)$ and $i(t)$ related in the circuit below??



A: It turns out that **any** circuit constructed **entirely** with linear circuit elements is **likewise** a linear system (i.e., a linear circuit).

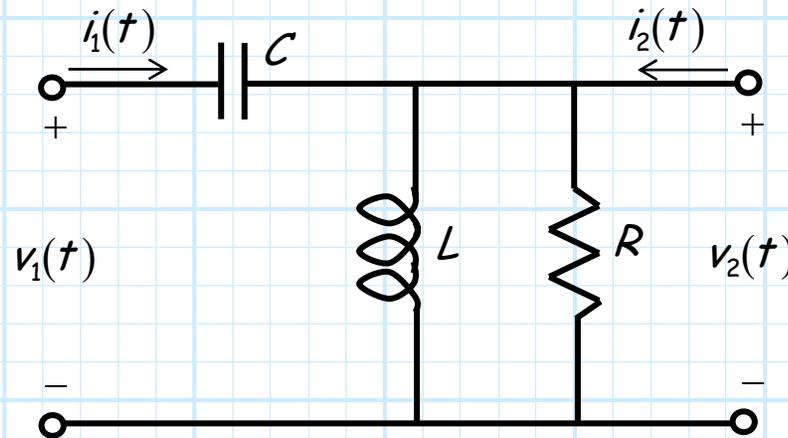
As a result, we know that that there **must** be some linear operator that relates $v(t)$ and $i(t)$ in your example!

$$\mathcal{L}_Z[i(t)] = v(t)$$

This is very useful for multi-port networks

The circuit above provides a good example of a **single-port** (a.k.a. **one-port**) network.

We can of course construct networks with **two or more** ports; an example of a **two-port network** is shown below:



Since this circuit is **linear**, the relationship between **all** voltages and currents can likewise be expressed as **linear operators**, e.g.:

$$\mathcal{L}_{z_{21}}[v_1(t)] = v_2(t) \quad \mathcal{L}_{z_{21}}[i_1(t)] = v_2(t) \quad \mathcal{L}_{z_{22}}[i_2(t)] = v_2(t)$$

The linear operator is a convolution integral

Q: *Yikes! What would these linear operators for this circuit be? How can we determine them?*

A: It turns out that linear operators for **all** linear circuits can all be expressed in precisely the **same** form!

For example, the linear operators of a single-port network are:

$$v(t) = \mathcal{L}_z[i(t)] = \int_{-\infty}^t g_z(t-t') i(t') dt'$$

$$i(t) = \mathcal{L}_y[v(t)] = \int_{-\infty}^t g_y(t-t') v(t') dt'$$



In other words, the linear operator of linear circuits can always be expressed as a **convolution** integral—a convolution with a **circuit impulse function** $g(t)$.

The impulse response

Q: *But just what is this "circuit impulse response"??*

A: An impulse response is simply the **response** of one circuit function (i.e., $i(t)$ or $v(t)$) due to a **specific** stimulus by another.



That specific stimulus is the **impulse function** $\delta(t)$.

The impulse function **can** be defined as:

$$\delta(t) = \lim_{\tau \rightarrow 0} \frac{1}{\tau} \frac{\sin\left(\frac{\pi t}{\tau}\right)}{\left(\frac{\pi t}{\tau}\right)}$$

Such that it has the following two **properties**:

1. $\delta(t) = 0$ for $t \neq 0$

2. $\int_{-\infty}^{\infty} \delta(t) dt = 1.0$

We can define all sorts of impulse responses

The impulse responses of the **one-port example** are therefore defined as:

$$g_z(t) \doteq v(t) \Big|_{i(t)=\delta(t)}$$

and:

$$g_y(t) \doteq i(t) \Big|_{v(t)=\delta(t)}$$



Meaning simply that $g_z(t)$ is equal to the **voltage** function $v(t)$ when the circuit is "thumped" with a **impulse current** (i.e., $i(t) = \delta(t)$), and $g_y(t)$ is equal to the **current** $i(t)$ when the circuit is "thumped" with a **impulse voltage** (i.e., $v(t) = \delta(t)$).

We can make convolution integrals simple!

Similarly, the relationship between the **input** and the **output** of a **two-port** network can be expressed as:

$$v_2(t) = \mathcal{L}_{21}[v_1(t)] = \int_{-\infty}^t g(t-t') v_1(t') dt'$$

where:

$$g(t) \doteq v_2(t) \Big|_{v_1(t)=\delta(t)}$$

Note that the circuit impulse response must be **causal** (nothing can occur at the output **until** something occurs at the input), so that:

$$g(t) = 0 \quad \text{for} \quad t < 0$$

Q: *Yikes! I recall evaluating convolution integrals to be messy, difficult and stressful. Surely there is an **easier** way to describe linear circuits!?!*

A: Nope! The convolution integral is **all** there is.

However, we can use our linear systems theory toolbox to greatly **simplify the evaluation** of a convolution integral!

The Eigen Function of Linear Systems

Recall that that we can express (**expand**) a time-limited signal with a weighted summation of **basis functions**:

$$v(t) = \sum_n a_n \psi_n(t)$$

where $v(t) = 0$ for $t < 0$ and $t > T$.

Say now that we **convolve** this signal with some system **impulse function** $g(t)$:

$$\begin{aligned} \mathcal{L}[v(t)] &= \int_{-\infty}^t g(t-t') v(t') dt' \\ &= \int_{-\infty}^t g(t-t') \sum_n a_n \psi_n(t') dt' \\ &= \sum_n a_n \int_{-\infty}^t g(t-t') \psi_n(t') dt' \end{aligned}$$

Look what happened!

Convolve with the basis functions - not the signal

Instead of convolving the general function $v(t)$, we now find that we must simply convolve with the set of basis functions $\psi_n(t)$.

Q: *Huh? You say we must "simply" convolve the set of basis functions $\psi_n(t)$. Why would this be any simpler?*

A: Remember, **you** get to **choose** the basis $\psi_n(t)$. If you're **smart**, you'll choose a set that makes the convolution integral "**simple**" to perform!

Q: *But don't I first need to know the explicit form of $g(t)$ before I intelligently choose $\psi_n(t)$??*

A: Not necessarily!

Time to use our "special" basis

The **key** here is that the convolution integral:

$$\mathcal{L}[\psi_n(t)] = \int_{-\infty}^t g(t-t') \psi_n(t') dt'$$

is a **linear, time-invariant** operator.

Because of this, there exists one **basis** with an **astonishing** property!

These **special** basis functions are:



$$\psi_n(t) = \begin{cases} e^{j\omega_n t} & \text{for } 0 \leq t \leq T \\ 0 & \text{for } t < 0, t > T \end{cases}$$

where $\omega_n = n \left(\frac{2\pi}{T} \right)$

Prof. Stiles: So darn lame

Now, inserting this function (get ready, here comes the **astonishing** part!) into the convolution integral:

$$\mathcal{L}[e^{j\omega_n t}] = \int_{-\infty}^t g(t-t') e^{j\omega_n t'} dt'$$

and using the substitution $u = t - t'$, we get:

$$\begin{aligned} \int_{-\infty}^t g(t-t') e^{j\omega_n t'} dt' &= \int_{t-(-\infty)}^{t-t} g(u) e^{j\omega_n(t-u)} (-du) \\ &= e^{j\omega_n t} \int_{+\infty}^0 g(u) e^{-j\omega_n u} (-du) \\ &= e^{j\omega_n t} \int_0^{\infty} g(u) e^{-j\omega_n u} du \end{aligned}$$



See! Doesn't **that** astonish!

Q: *I'm only astonished by how **lame** you are. How is this result any **more** "astonishing" than any of the **other** "useful" things you've been telling us?*

Convolution becomes multiplication

A: Note that the integration in this **result** is **not** a convolution—the integral is simply a **value** that depends on n (but **not** time t):

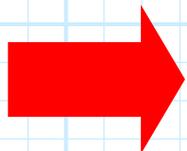
$$G(\omega_n) \doteq \int_0^{\infty} g(t) e^{-j\omega_n t} dt$$

As a result, convolution with this “special” set of basis functions can **always** be expressed as:

$$\int_{-\infty}^t g(t-t') e^{j\omega_n t'} dt' = \mathcal{L}[e^{j\omega_n t}] = G(\omega_n) e^{j\omega_n t}$$

The **remarkable** thing about this result is that the linear operation on function $\psi_n(t) = \exp[j\omega_n t]$ results in precisely the **same** function of **time** t (save the **complex** multiplier $G(\omega_n)$)! I.E.:

$$\mathcal{L}[\psi_n(t)] = G(\omega_n) \psi_n(t)$$



Convolution with $\psi_n(t) = \exp[j\omega_n t]$ is accomplished by simply **multiplying** the function by the **complex** number $G(\omega_n)$!

This only works for complex exponentials

Note this is true **regardless** of the impulse response $g(t)$ (the function $g(t)$ affects the **value** of $G(\omega_n)$ **only**!)

Q: *Big deal! Aren't there lots of **other** functions that would satisfy the equation above equation?*

A: Nope. The **only** function where this is true is:

$$\psi_n(t) = e^{j\omega_n t}$$

This function is thus **very** special.

We call this function the **eigen function** of linear, time-invariant systems.

But complex exponentials are two sinusoidal functions

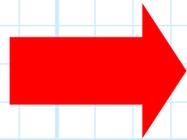
Q: *Are you **sure** that there are no other Eigen functions??*

A: Well, sort of.

Recall from **Euler's equation** that:

$$e^{j\omega_n t} = \cos \omega_n t + j \sin \omega_n t$$

It can be shown that the sinusoidal functions $\cos \omega_n t$ and $\sin \omega_n t$ are **likewise** Eigen functions of linear, time-invariant systems.



The real and imaginary components of Eigen function $\exp[j\omega_n t]$ are **also** Eigen functions.

Every linear operator has its Eigen value

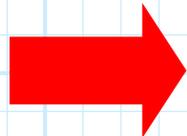
Q: *What about the set of values $G(\omega_n)$?? Do they have any significance or importance??*

A: Absolutely!

Recall the values $G(\omega_n)$ (one for each n) depend on the **impulse response** of the system (e.g., circuit) **only**:

$$G(\omega_n) \doteq \int_0^{\infty} g(t) e^{-j\omega_n t} dt$$

Thus, the set of values $G(\omega_n)$ completely **characterizes** a linear time-invariant **circuit** over time $0 \leq t \leq T$.



We call the values $G(\omega_n)$ the **Eigen values** of the linear, time-invariant circuit.

We're electrical engineers: why should we care?

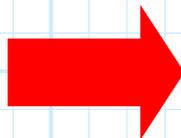


Q: *OK Poindexter, all **Eigen** stuff this **might** be interesting if you're a mathematician, but is it at all **useful** to us **electrical engineers**?*

A: It is **unfathomably** useful to us electrical engineers!

Say a linear, time-invariant circuit is excited (only) by a **sinusoidal** source (e.g., $v_s(t) = \cos \omega_o t$).

Since the source function is the **Eigen function** of the circuit, we will find that at **every** point in the circuit, **both** the current and voltage will have the **same functional form**.

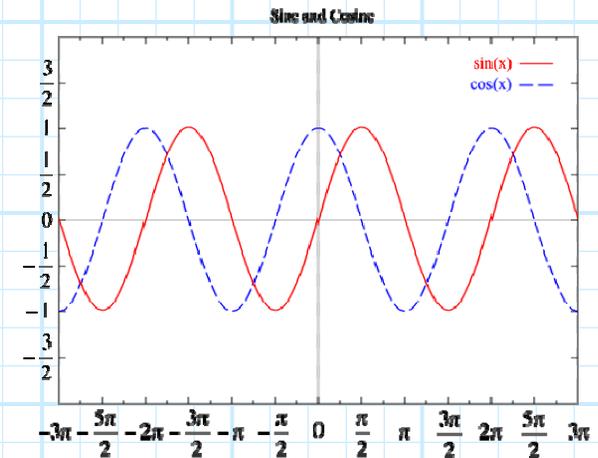
 That is, every current and voltage in the circuit will likewise be a **perfect sinusoid** with frequency ω_o !!

Haven't you wondered why we always use these?

Of course, the **magnitude** of the sinusoidal oscillation will be **different** at different points within the circuit, as will the **relative phase**.

But we know that **every** current and voltage in the circuit can be **precisely** expressed as a function of this form:

$$A \cos(\omega_o t + \varphi)$$



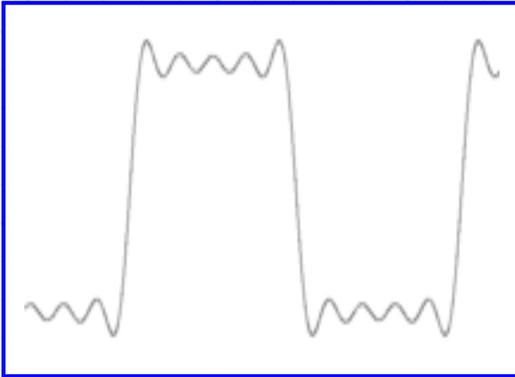
Q: *Isn't this pretty obvious?*

A: Why should it be?

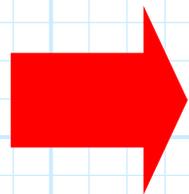
Say our source function was instead a **square** wave, or **triangle** wave, or a **sawtooth** wave.

We would find that (generally speaking) **nowhere** in the circuit would we find another current or voltage that was a **perfect** square wave (etc.)!

We “just” have to determine magnitude and phase!



In fact, we would find that not only are the current and voltage functions within the circuit **different** than the source function (e.g. a sawtooth) they are (generally speaking) all different **from each other**.



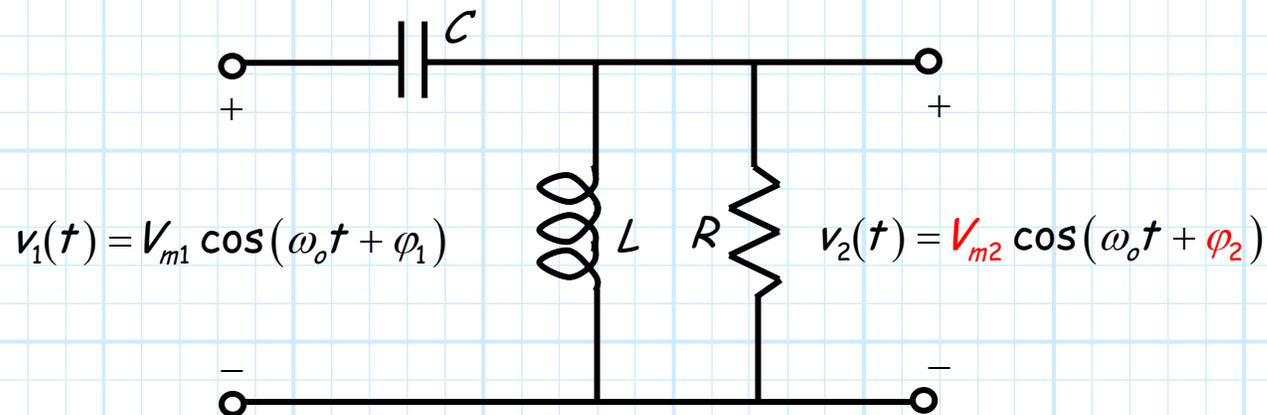
We find then that a linear circuit will (generally speaking) **distort** any source function—**unless** that function is the **Eigen function** (i.e., a sinusoidal function).

Thus, using an **Eigen function** as circuit source greatly simplifies our linear circuit analysis problem.

All we need to accomplish this is to **determine the magnitude A and relative phase ϕ** of the resulting (and otherwise **identical**) sinusoidal function!

A Complex Representation of Sinusoidal Functions

Q: *So, you say (for example) if a linear two-port circuit is driven by a sinusoidal source with arbitrary frequency ω_o , then the output will be identically sinusoidal, only with a different magnitude and relative phase.*



How do we determine the unknown magnitude V_{m2} and phase φ_2 of this output?

Eigen values are complex

A: Say the input and output are related by the impulse response $g(t)$:

$$v_2(t) = \mathcal{L}[v_1(t)] = \int_{-\infty}^t g(t-t') v_1(t') dt'$$

We now know that **if** the input were **instead**:

$$v_1(t) = e^{j\omega_0 t}$$

then:

$$v_2(t) = \mathcal{L}[e^{j\omega_0 t}] = G(\omega_0) e^{j\omega_0 t}$$

where:

$$G(\omega_0) \doteq \int_0^{\infty} g(t) e^{-j\omega_0 t} dt$$

Thus, we simply multiply the input $v_1(t) = e^{j\omega_0 t}$ by the **complex** eigen value $G(\omega_0)$ to determine the **complex** output $v_2(t)$:

$$v_2(t) = G(\omega_0) e^{j\omega_0 t}$$

Complex voltages and currents are your friend!



Q: *You professors drive me crazy with all this math involving **complex** (i.e., real and imaginary) voltage functions. In the lab I can only generate and measure **real-valued** voltages and **real-valued** voltage functions. Voltage is a **real-valued, physical** parameter!*

A: You are quite **correct**.

Voltage is a real-valued parameter, expressing electric potential (in Joules) per unit charge (in Coulombs).

Q: *So, all your **complex** formulations and **complex** eigen values and **complex** eigen functions may all be sound **mathematical abstractions**, but aren't they **worthless** to us **electrical engineers** who work in the "**real**" world (pun intended)?*

A: Absolutely not! Complex analysis actually **simplifies** our analysis of real-valued voltages and currents in **linear circuits** (but **only** for linear circuits!).

Remember Euler

The key relationship comes from **Euler's Identity**:

$$e^{j\omega t} = \cos \omega t + j \sin \omega t$$

Meaning:

$$\operatorname{Re}\{e^{j\omega t}\} = \cos \omega t$$



Now, consider a **complex value** C . We of course can write this complex number in terms of its **real** and **imaginary** parts:

$$C = a + j b \quad \therefore \quad a = \operatorname{Re}\{C\} \quad \text{and} \quad b = \operatorname{Im}\{C\}$$

But, we can **also** write it in terms of its **magnitude** $|C|$ and **phase** φ !

$$C = |C| e^{j\varphi}$$

where:

$$|C| = C C^* = a^2 + b^2 \quad \varphi = \tan^{-1} \left[\frac{b}{a} \right]$$

A complex number has magnitude and phase

Thus, the complex function $C e^{j\omega_0 t}$ is:

$$\begin{aligned} C e^{j\omega_0 t} &= |C| e^{j\varphi} e^{j\omega_0 t} \\ &= |C| e^{j\omega_0 t + \varphi} \\ &= |C| \cos(\omega_0 t + \varphi) + j |C| \sin(\omega_0 t + \varphi) \end{aligned}$$

Therefore we find:

$$|C| \cos(\omega_0 t + \varphi) = \operatorname{Re} \{ C e^{j\omega_0 t} \}$$

Now, consider again the **real-valued** voltage function:

$$v_1(t) = V_{m1} \cos(\omega t + \varphi_1)$$

This function is of course **sinusoidal** with a magnitude V_{m1} and phase φ_1 .

Using what we have learned above, we can **likewise** express this real function as:

$$v_1(t) = V_{m1} \cos(\omega t + \varphi_1) = \operatorname{Re} \{ V_1 e^{j\omega t} \}$$

where V_1 is the **complex number**: $V_1 = V_{m1} e^{j\varphi_1}$

But what is the output signal?

Q: *I see! A real-valued sinusoid has a magnitude and phase, just like complex number.*

A single complex number (V) can be used to specify both of the fundamental (real-valued) parameters of our sinusoid (V_m, φ).

What I don't see is how this helps us in our circuit analysis.

After all:

$$v_2(t) \neq G(\omega_o) \operatorname{Re}\{V_1 e^{j\omega_o t}\}$$

What then is the real-valued output $v_2(t)$ of our two-port network when the input $v_1(t)$ is the real-valued sinusoid:

$$\begin{aligned} v_1(t) &= V_{m1} \cos(\omega_o t + \varphi_1) \\ &= \operatorname{Re}\{V_1 e^{j\omega_o t}\} \end{aligned} \quad ???$$

The math will reveal the answer!

A: Let's go back to our **original** convolution integral:

$$v_2(t) = \int_{-\infty}^t g(t-t') v_1(t') dt'$$

If:

$$\begin{aligned} v_1(t) &= V_{m1} \cos(\omega_o t + \phi_1) \\ &= \operatorname{Re} \{ V_1 e^{j\omega_o t} \} \end{aligned}$$

then:

$$v_2(t) = \int_{-\infty}^t g(t-t') \operatorname{Re} \{ V_1 e^{j\omega_o t'} \} dt'$$

Now, since the impulse function $g(t)$ is **real-valued** (this is really important!) it can be shown that:

$$\begin{aligned} v_2(t) &= \int_{-\infty}^t g(t-t') \operatorname{Re} \{ V_1 e^{j\omega_o t'} \} dt' \\ &= \operatorname{Re} \left\{ \int_{-\infty}^t g(t-t') V_1 e^{j\omega_o t'} dt' \right\} \end{aligned}$$

The output signal

Now, applying what we have previously learned;

$$\begin{aligned} v_2(t) &= \operatorname{Re} \left\{ \int_{-\infty}^t g(t-t') V_1 e^{j\omega_0 t'} dt' \right\} \\ &= \operatorname{Re} \left\{ V_1 \int_{-\infty}^t g(t-t') e^{j\omega_0 t'} dt' \right\} \\ &= \operatorname{Re} \left\{ V_1 G(\omega_0) e^{j\omega_0 t} \right\} \end{aligned}$$

Thus, we **finally** can conclude the real-valued output $v_2(t)$ due to the **real-valued** input:

$$v_1(t) = V_{m1} \cos(\omega_0 t + \varphi_1) = \operatorname{Re} \left\{ V_1 e^{j\omega_0 t} \right\}$$

is:

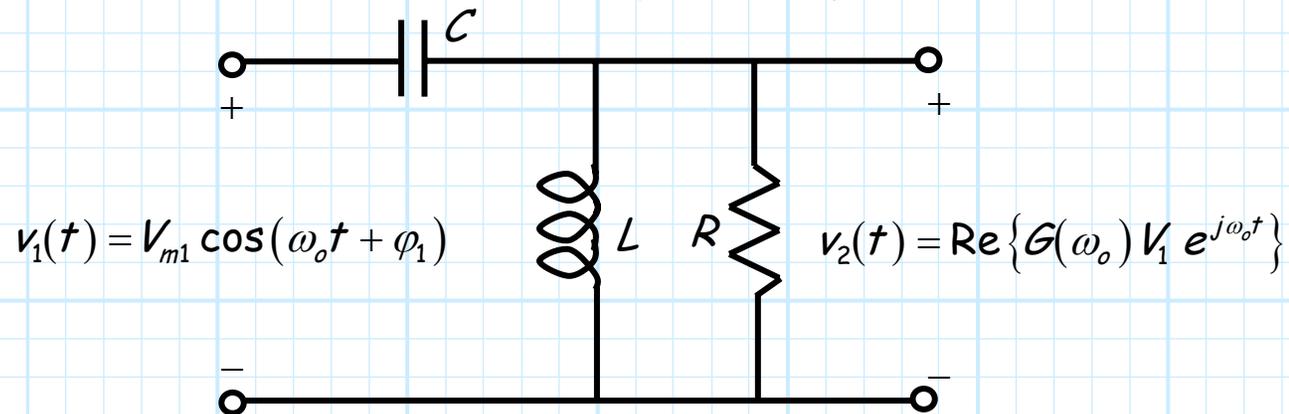
$$v_2(t) = \operatorname{Re} \left\{ V_2 e^{j\omega_0 t} \right\} = V_{m2} \cos(\omega_0 t + \varphi_2)$$

where:

$$V_2 = G(\omega_0) V_1$$

The **really important** result here is the last one!

The Eigen value of the Linear operator is its "Frequency Response"



The magnitude and phase of the **output** sinusoid (expressed as **complex** value V_2) is related to the magnitude and phase of the **input** sinusoid (expressed as **complex** value V_1) by the system **eigen value** $G(\omega_o)$:

$$\frac{V_2}{V_1} = G(\omega_o)$$

Therefore we find that **really** often in electrical engineering, we:

1. Use sinusoidal (i.e., eigen function) sources.
2. Express the voltages and currents created by these sources as complex values (i.e., not as real functions of time)!

Make sure you know what complex voltages and currents physically represent!

For **example**, we might say " $V_3 = 2.0$ ", meaning:

$$V_3 = 2.0 = 2.0 e^{j0} \Rightarrow v_3(t) = \operatorname{Re} \{ 2.0 e^{j0} e^{j\omega_0 t} \} = 2.0 \cos \omega_0 t$$

Or " $I_L = -3.0$ ", meaning:

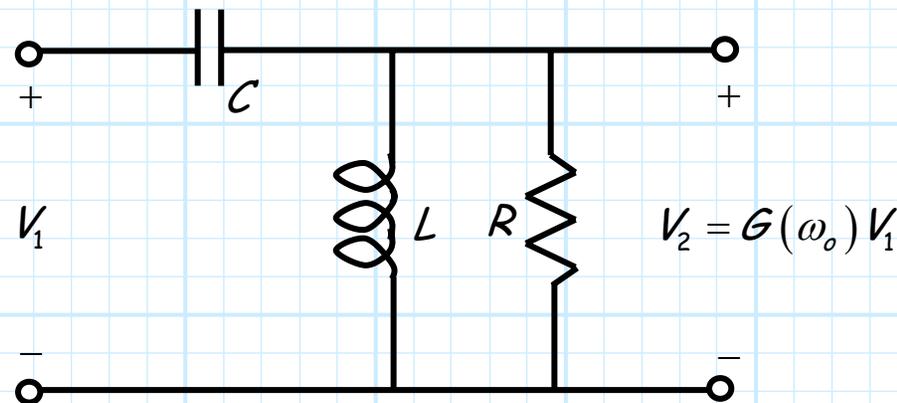
$$I_L = -3.0 = 3.0 e^{j\pi} \Rightarrow i_L(t) = \operatorname{Re} \{ 3.0 e^{j\pi} e^{j\omega_0 t} \} = 3.0 \cos(\omega_0 t + \pi)$$

Or " $V_s = j$ ", meaning:

$$V_s = j = 1.0 e^{j(\pi/2)} \Rightarrow v_s(t) = \operatorname{Re} \{ 1.0 e^{j(\pi/2)} e^{j\omega_0 t} \} = 1.0 \cos(\omega_0 t + \pi/2)$$

Summarizing

- * Remember, if a linear circuit is excited by a sinusoid (e.g., **eigen function** $\exp[j\omega_0 t]$), then the **only** unknowns are the magnitude and phase of the sinusoidal **currents** and **voltages** associated with **each element** of the circuit.
- * These unknowns are **completely** described by complex values, as complex values **likewise** have a magnitude and phase.
- * We can always **"recover"** the **real-valued** voltage or current function by multiplying the complex value by $\exp[j\omega_0 t]$ and then taking the real part, but typically we don't—after all, **no** new or unknown information is revealed by this operation!



Analysis of Circuits Driven by Arbitrary Functions

Q: *What happens if a linear circuit is excited by some function that is **not** an "eigen function"? Isn't limiting our analysis to sinusoids **too restrictive**?*

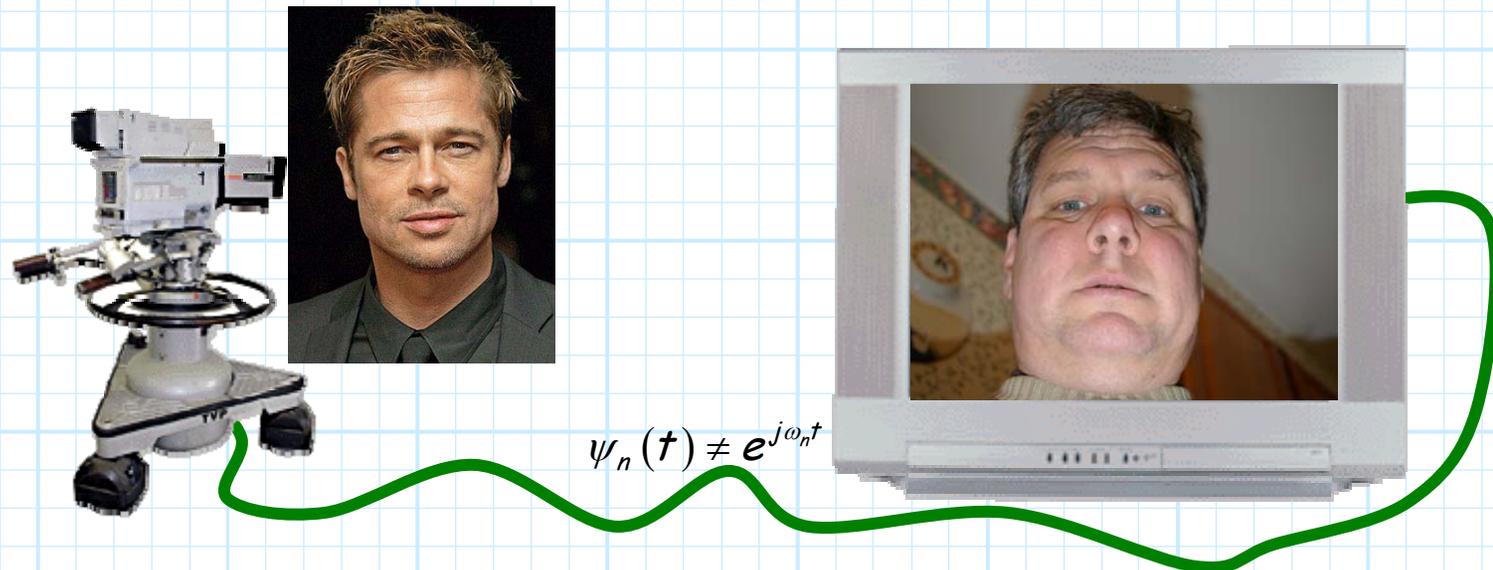
A: Not as restrictive as you might think.

Because sinusoidal functions are the eigen-functions of linear, time-invariant systems, they have become **fundamental** to much of our electrical engineering infrastructure—particularly with regard to **communications**.

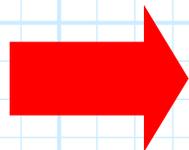
For example, every radio and TV station is assigned its **very own eigen function** (i.e., its own frequency ω)!

Eigen functions: without them communication would be impossible

It is **very** important that we use eigen functions for electromagnetic communication, otherwise the **received** signal might look **grotesquely** different from the one that was **transmitted!**



With sinusoidal functions (being eigen functions and all), we **know** that receive function will have **precisely** the same form as the one transmitted (albeit quite a bit **smaller**).



Thus, our assumption that a linear circuit is excited by a sinusoidal function is often a very **accurate** and **practical** one!

What if the signal is not sinusoidal?

Q: *Still, we often find a circuit that is **not** driven by a sinusoidal source. How would we analyze **this** circuit?*

A: Recall the property of **linear operators**:

$$\mathcal{L}[a y_1 + b y_2] = a \mathcal{L}[y_1] + b \mathcal{L}[y_2]$$

We now know that we can **expand** the function:

$$v(t) = a_0 \psi_0(t) + a_1 \psi_1(t) + a_2 \psi_2(t) + \dots = \sum_{n=-\infty}^{\infty} a_n \psi_n(t)$$

and we found that:

$$\mathcal{L}[v(t)] = \mathcal{L}\left[\sum_{n=-\infty}^{\infty} a_n \psi_n(t)\right] = \sum_{n=-\infty}^{\infty} a_n \mathcal{L}[\psi_n(t)]$$

Let's choose Eigen functions as our basis

We found that any linear operation $\mathcal{L}[\psi_n(t)]$ is greatly simplified if we choose as our basis function the **eigen function** of linear systems:

$$\psi_n(t) = \begin{cases} e^{j\omega_n t} & \text{for } 0 \leq t \leq T \\ 0 & \text{for } t < 0, t > T \end{cases} \quad \text{where} \quad \omega_n = n \left(\frac{2\pi}{T} \right)$$

so that:

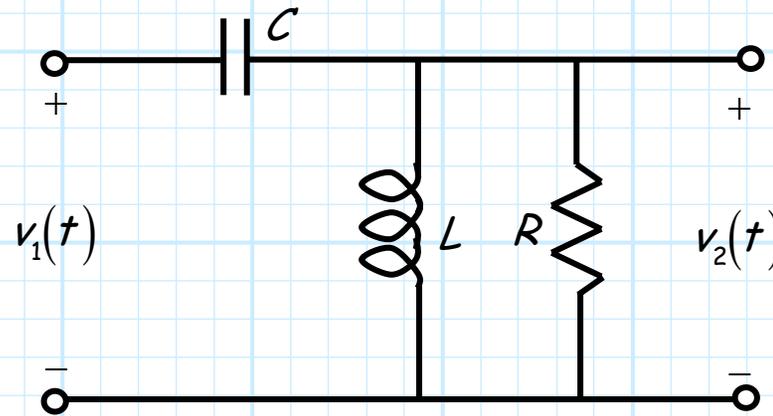
$$\mathcal{L}[\psi_n(t)] = G(\omega_n) e^{j\omega_n t}$$

And so:

$$\mathcal{L}[v(t)] = \mathcal{L}\left[\sum_{n=-\infty}^{\infty} a_n e^{j\omega_n t} \right] = \sum_{n=-\infty}^{\infty} a_n \mathcal{L}\left[e^{j\omega_n t} \right] = \sum_{n=-\infty}^{\infty} a_n G(\omega_n) e^{j\omega_n t}$$

Just follow these steps...

Thus, for the example:



We follow these analysis steps:

1. Expand the input function $v_1(t)$ using the basis functions $\psi_n(t) = \exp[j\omega_n t]$:

$$v_1(t) = V_{01} e^{j\omega_0 t} + V_{11} e^{j\omega_1 t} + V_{21} e^{j\omega_2 t} + \dots = \sum_{n=-\infty}^{\infty} V_{n1} e^{j\omega_n t}$$

where:

$$V_{n1} = \frac{1}{T} \int_0^T v_1(t) e^{-j\omega_n t} dt$$

...and the output is determined

2. Evaluate the **eigen values** of the linear system:

$$G(\omega_n) = \int_0^{\infty} g(t) e^{-j\omega_n t} dt$$

3. Perform the **linear operation** (the convolution integral) that relates $v_2(t)$ to $v_1(t)$:

$$\begin{aligned} v_2(t) &= \mathcal{L}[v_1(t)] \\ &= \mathcal{L}\left[\sum_{n=-\infty}^{\infty} V_{n1} e^{j\omega_n t}\right] \\ &= \sum_{n=-\infty}^{\infty} V_{n1} \mathcal{L}[e^{j\omega_n t}] \\ &= \sum_{n=-\infty}^{\infty} V_{n1} G(\omega_n) e^{j\omega_n t} \end{aligned}$$

A Summary

Summarizing:

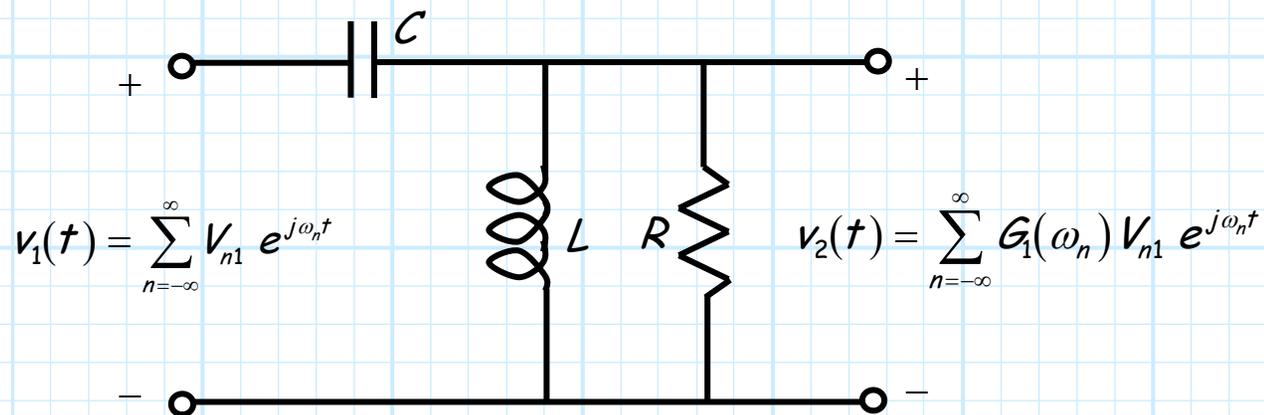
$$v_2(t) = \sum_{n=-\infty}^{\infty} V_{n2} e^{j\omega_n t}$$

where:

$$V_{n2} = G(\omega_n) V_{n1}$$

and:

$$V_{n1} = \frac{1}{T} \int_0^T v_1(t) e^{-j\omega_n t} dt \quad G(\omega_n) = \int_0^{\infty} g(t) e^{-j\omega_n t} dt$$



As stated earlier, the signal expansion used here is the **Fourier Series**.

The Fourier Transform

Say that the **timewidth** T of the signal $v_1(t)$ becomes **infinite**. In this case we find our analysis becomes:

$$v_2(t) = \frac{1}{2\pi} \int_{-\infty}^{+\infty} V_2(\omega) e^{j\omega t} d\omega$$

where:

$$V_2(\omega) = \mathcal{G}(\omega) V_1(\omega)$$

and:

$$V_1(\omega) = \int_{-\infty}^{+\infty} v_1(t) e^{-j\omega t} dt \quad \mathcal{G}(\omega) = \int_{-\infty}^{+\infty} g(t) e^{-j\omega t} dt$$

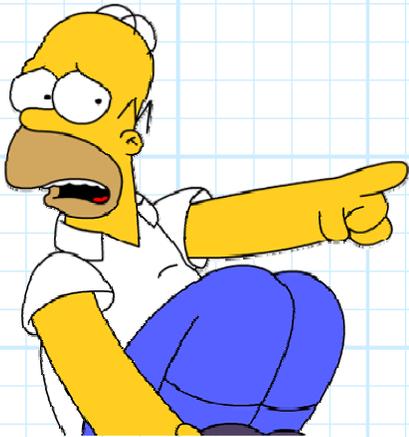
The signal expansion in this case is the **Fourier Transform**.

We find that as $T \rightarrow \infty$ the number of **discrete** system eigen values $\mathcal{G}(\omega_n)$ become so numerous that they form a **continuum**— $\mathcal{G}(\omega)$ is a **continuous** function of frequency ω .

We thus call the function $\mathcal{G}(\omega)$ the **eigen spectrum** or **frequency response** of the circuit.

This still looks very difficult!

Q: *You claim that all this fancy mathematics (e.g., eigen functions and eigen values) make analysis of linear systems and circuits much **easier**, yet to apply these techniques, we must **determine** the eigen values or eigen spectrum:*



$$G(\omega_n) = \int_0^{\infty} g(t) e^{-j\omega_n t} dt \quad G(\omega) = \int_{-\infty}^{+\infty} g(t) e^{-j\omega t} dt$$

Neither of these operations look at all easy.

*And in addition to performing the integration, we must **somehow** determine the **impulse function** $g(t)$ of the linear system as well!*

*Just how are we supposed to do **that**?*

It's not nearly as difficult as it appears!

A: An insightful question!

Determining the impulse response $g(t)$ and then the frequency response $G(\omega)$ **does** appear to be **exceedingly** difficult—and for many linear systems it indeed **is!**

However, much to our great **relief**, we can determine the eigen spectrum $G(\omega)$ of linear circuits **without** having to perform a difficult integration.

In fact, we **don't** even need to know the impulse response $g(t)$!



The Eigen Values of Linear Circuits

Recall the linear operators that define a capacitor:

$$\mathcal{L}_y^c[v_c(t)] = i_c(t) = C \frac{dv_c(t)}{dt}$$

$$\mathcal{L}_z^c[i_c(t)] = v_c(t) = \frac{1}{C} \int_{-\infty}^t i_c(t') dt'$$

We now know that the **Eigen function** of these linear, time-invariant operators—like **all** linear, time-invariant operators—is $\exp[j\omega t]$.

The question now is: **what is the Eigen value** of each of these operators?

It is this value that **defines** the physical behavior of a given capacitor!

The operator is linear

For $v_c(t) = \exp[j\omega t]$, we find:

$$\begin{aligned}i_c(t) &= \mathcal{L}_y^c[v_c(t)] \\ &= C \frac{d e^{j\omega t}}{dt} \\ &= (j\omega C) e^{j\omega t}\end{aligned}$$

Just as we expected, the Eigen function $\exp[j\omega t]$ “survives” the linear operation **unscathed**—the current function $i(t)$ has **precisely** the same form as the voltage function $v(t) = \exp[j\omega t]$.

The **only** difference between the **current** and **voltage** is the multiplication of the **Eigen value**, denoted as $G_y^c(\omega)$.

$$i_c(t) = \mathcal{L}_y^c[v(t) = e^{j\omega t}] = G_y^c(\omega) e^{j\omega t}$$

The Eigen value of a capacitor

Since we **just** determined that for this case:

$$i_c(t) = (j\omega C) e^{j\omega t}$$

it is **evident** that the Eigen value of the linear operation:

$$i(t) = \mathcal{L}_y^C[v(t)] = C \frac{dv(t)}{dt}$$

is:

$$G_y^C(\omega) = j\omega C = \omega C e^{j\pi/2} \quad !!!$$

Let's now consider real-valued functions

So for example, if:

$$\begin{aligned} v(t) &= V_m \cos(\omega_o t + \varphi) \\ &= \operatorname{Re} \left\{ (V_m e^{j\varphi}) e^{j\omega_o t} \right\} \end{aligned}$$

we will find that:

$$\begin{aligned} \mathcal{L}_y^{\mathcal{C}} \left[(V_m e^{j\varphi}) e^{j\omega_o t} \right] &= \mathcal{G}_y^{\mathcal{C}}(\omega_o) (V_m e^{j\varphi}) e^{j\omega_o t} \\ &= \left(\omega C e^{j\pi/2} \right) (V_m e^{j\varphi}) e^{j\omega_o t} \\ &= \left(\omega C V_m e^{j(\pi/2 + \varphi)} \right) e^{j\omega_o t} \end{aligned}$$

Therefore:

$$\begin{aligned} i_c(t) &= \operatorname{Re} \left\{ \omega C V_m e^{j(\varphi + \pi/2)} e^{j\omega_o t} \right\} \\ &= \omega C V_m \cos\left(\omega_o t + \varphi + \pi/2\right) \\ &= -\omega C V_m \sin(\omega_o t + \varphi) \end{aligned}$$

Remember what the complex value means

Hopefully, this example again emphasizes that these **real-valued** sinusoidal functions can be completely expressed in terms of **complex values**.

For **example**, the complex value:

$$V_C = V_m e^{j\varphi}$$

means that the magnitude of the sinusoidal **voltage** is $|V_C| = V_m$, and its relative phase is $\angle V_C = \varphi$. The complex value:

$$I_C = G_y^C(\omega) V_C = \left(\omega C e^{j\pi/2} \right) V_C$$

likewise means that the **magnitude** of the sinusoidal **current** is:

$$|I_C| = |G_y^C(\omega) V_C| = |G_y^C(\omega)| |V_C| = \omega C V_m$$

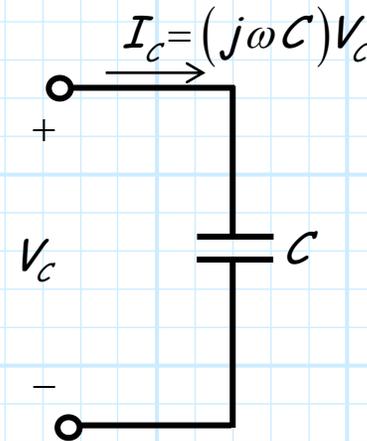
And the relative **phase** of the sinusoidal **current** is:

$$\angle I_C = \angle G_y^C(\omega) + \angle V_C = \frac{\pi}{2} + \varphi$$

Now find the voltage from the current

We can thus **summarize** the behavior of a capacitor with the simple **complex equation**:

$$\begin{aligned} I_c &= (j\omega C) V_c \\ &= \left(\omega C e^{j\pi/2} \right) V_c \end{aligned}$$



Now let's return to the **second** of the two linear operators that describe a capacitor:

$$v_c(t) = \mathcal{L}_z^C [i_c(t)] = \frac{1}{C} \int_{-\infty}^t i_c(t') dt'$$

Now, if the capacitor **current** is the Eigen function $i_c(t) = \exp[j\omega t]$, we find:

$$\mathcal{L}_z^C [e^{j\omega t}] = \frac{1}{C} \int_{-\infty}^t e^{j\omega t'} dt' = \left(\frac{1}{j\omega C} \right) e^{j\omega t}$$

where we assume $i(t = -\infty) = 0$.

The Eigen value of this linear operator

Thus, we can conclude that:

$$\mathcal{L}_Z^C [e^{j\omega t}] = G_Z^C(\omega) e^{j\omega t} = \left(\frac{1}{j\omega C} \right) e^{j\omega t}$$

Hopefully, it is evident that the **Eigen value** of this linear operator is:

$$G_Z^C(\omega) = \frac{1}{j\omega C} = \frac{-j}{\omega C} = \frac{1}{\omega C} e^{j(3\pi/2)}$$

And so:

$$V_C = \left(\frac{1}{j\omega C} \right) I_C$$

Impedance is simply an Eigen value!

Q: *Wait a second! Isn't this essentially the **same** result as the one derived for operator \mathcal{L}_y^C ??*

A: It's **precisely** the same! For both operators we find:

$$\frac{V_C}{I_C} = \frac{1}{j\omega C}$$

This should **not** be surprising, as **both** operators \mathcal{L}_y^C and \mathcal{L}_z^C relate the current through and voltage across the **same** device (a capacitor).

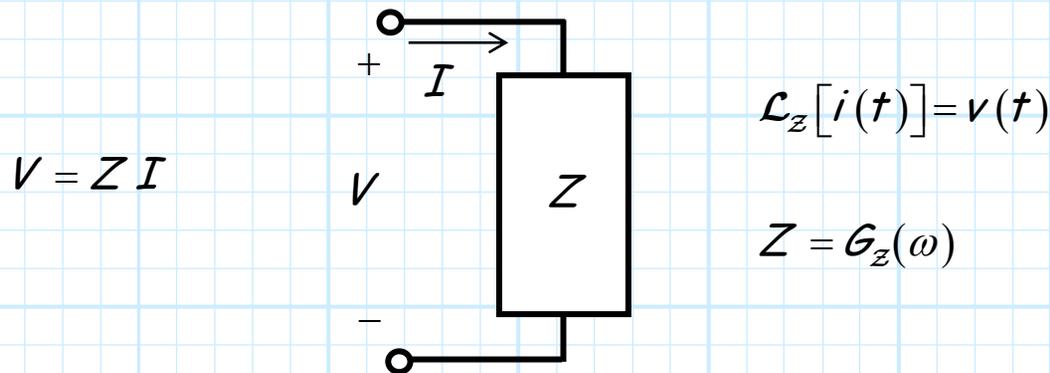
The **ratio** of complex voltage to complex current is of course referred to as the complex device **impedance** Z .

$$Z \doteq \frac{V}{I}$$

An **impedance** can be determined for **any** linear, time-invariant **one-port** network—but **only** for linear, time-invariant one-port networks!

Know what impedance tells you!

Generally speaking, impedance is a **function of frequency**. In fact, the impedance of a one-port network is simply the **Eigen value** $G_Z(\omega)$ of the linear operator \mathcal{L}_Z :



Note that impedance is a **complex** value that provides us with **two** things:

1. The **ratio of the magnitudes** of the sinusoidal voltage and current:

$$|Z| = \frac{|V|}{|I|}$$

2. The **difference in phase** between the sinusoidal voltage and current:

$$\angle Z = \angle V - \angle I$$

Admittance

Q: *What about the linear operator:*

$$\mathcal{L}_y[v(t)] = i(t) \quad ??$$

A: Hopefully it is now evident to **you** that:

$$G_y(\omega) = \frac{1}{G_z(\omega)} = \frac{1}{Z}$$

The inverse of impedance is **admittance** Y :

$$Y \doteq \frac{1}{Z} = \frac{I}{V}$$

Inductors and resistors

Now, returning to the **other two** linear circuit elements, we find (and **you** can verify) that for resistors:

$$\mathcal{L}_y^R[v_R(t)] = i_R(t) \Rightarrow G_y^R(\omega) = 1/R$$

$$\mathcal{L}_z^R[i_R(t)] = v_R(t) \Rightarrow G_z^R(\omega) = R$$

and for inductors:

$$\mathcal{L}_y^L[v_L(t)] = i_L(t) \Rightarrow G_y^L(\omega) = \frac{1}{j\omega L}$$

$$\mathcal{L}_z^L[i_L(t)] = v_L(t) \Rightarrow G_z^L(\omega) = j\omega L$$

meaning:

$$Z_R = \frac{1}{Y_R} = R = R e^{j0} \quad \text{and} \quad Z_L = \frac{1}{Y_L} = j\omega L = \omega L e^{j(\pi/2)}$$

All the rules of circuit theory apply to complex currents and voltages too

Now, note that the relationship

$$Z = \frac{V}{I}$$

forms a **complex "Ohm's Law"** with regard to complex currents and voltages.

Additionally, ICBST (It Can Be Shown That) **Kirchoff's Laws** are likewise valid for complex currents and voltages:

$$\sum_n I_n = 0 \qquad \sum_n V_n = 0$$

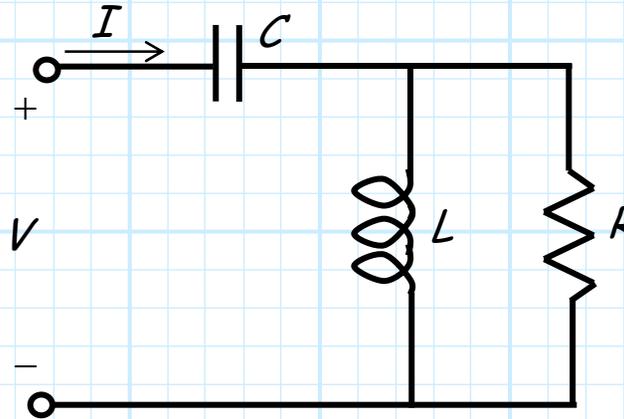
where of course the summation represents **complex addition**.

As a result, the impedance (i.e., the Eigen value) of **any** one-port device can be determined by simply applying a **basic** knowledge of **linear circuit analysis!**

We can determine Eigen values without knowing the impulse response!

Returning to the example:

$$Z = \frac{V}{I}$$



And thus using out **basic** circuits knowledge, we find:

$$Z = Z_C + Z_R \parallel Z_L = \frac{1}{j\omega C} + R \parallel j\omega L$$

Thus, the Eigen value of the linear operator:

$$\mathcal{L}_Z[i(t)] = v(t)$$

For **this** one-port network is:

$$G_Z(\omega) = \frac{1}{j\omega C} + R \parallel j\omega L$$

No need for convolution!

Look what we did! We were able to determine $G_z(\omega)$ **without** explicitly determining impulse response $g_z(t)$, or having to perform **any** integrations!

Now, if we actually **need** to determine the voltage function $v(t)$ created by some **arbitrary** current function $i(t)$, we integrate:

$$\begin{aligned} v(t) &= \frac{1}{2\pi} \int_{-\infty}^{+\infty} G_z(\omega) I(\omega) e^{j\omega t} d\omega \\ &= \frac{1}{2\pi} \int_{-\infty}^{+\infty} \left(\frac{1}{j\omega C} + R \parallel j\omega L \right) I(\omega) e^{j\omega t} d\omega \end{aligned}$$

where:

$$I(\omega) = \int_{-\infty}^{+\infty} i(t) e^{-j\omega t} dt$$

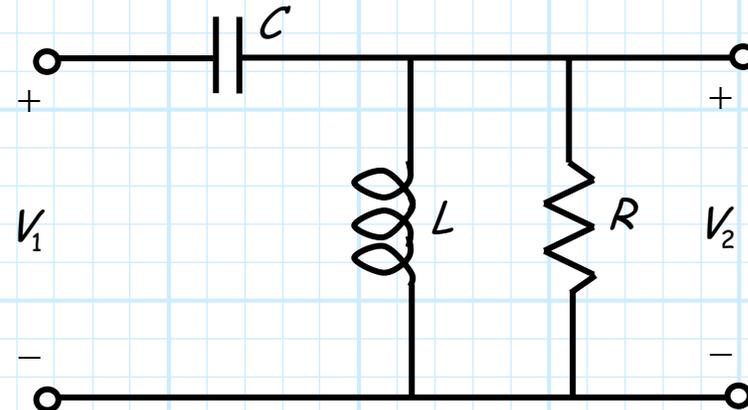
Otherwise, if our current function is **time-harmonic** (i.e., sinusoidal with frequency ω), we can simply relate complex current I and complex voltage V with the equation:

$$\begin{aligned} V &= Z I \\ &= \left(\frac{1}{j\omega C} + R \parallel j\omega L \right) I \end{aligned}$$

See how easy this is?

Similarly, for our **two-port** example, we can likewise determine from **basic** circuit theory the **Eigen value** of linear operator:

$$\mathcal{L}_{21}[v_1(t)] = v_2(t)$$



is:

$$G_{21}(\omega) = \frac{Z_L \parallel Z_R}{Z_C + Z_L \parallel Z_R} = \frac{j\omega L \parallel R}{\frac{1}{j\omega C} + j\omega L \parallel R}$$

so that:

$$V_2 = G_{21}(\omega) V_1$$

or more generally:

$$v_2(t) = \frac{1}{2\pi} \int_{-\infty}^{+\infty} G_{21}(\omega) V_1(\omega) e^{j\omega t} d\omega$$

where:

$$V_1(\omega) = \int_{-\infty}^{+\infty} v_1(t) e^{-j\omega t} dt$$

Eigen Values of the Laplace Transform

Well, I fibbed a little when I stated that the Eigen function of linear, time-invariant systems (circuits) is:

$$\mathcal{L}\{e^{j\omega t}\} = G(\omega)e^{j\omega t}$$



Instead, the more **general** Eigen function is:

$$\mathcal{L}\{e^{st}\} = G(s)e^{st}$$

Where s is a **complex** (i.e., real and imaginary) frequency of the form:

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

such that:

$$e^{st} = e^{(\sigma + j\omega)t} = e^{\sigma t} e^{j\omega t}$$

Note then, if $\sigma = 0$, the Eigen function e^{st} becomes the previously described Eigen function $e^{j\omega t}$!

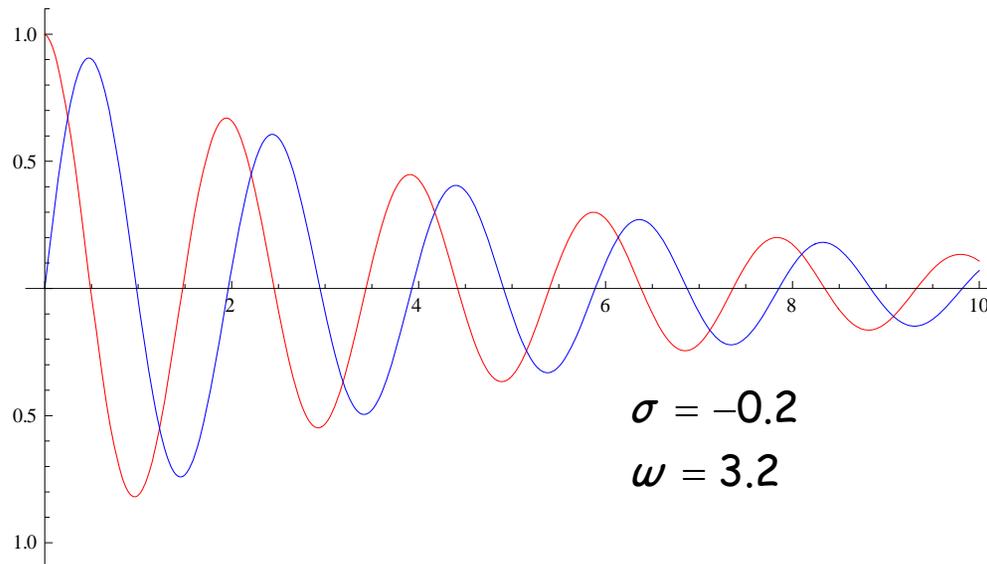
What does this function mean?

Q: *Yikes! I understand e^{st} even less than I understood $e^{j\omega t}$! What does this function mean?*

A: Remember, the function e^{st} is a **complex** function—it is actually an expression of **two real-value** functions.

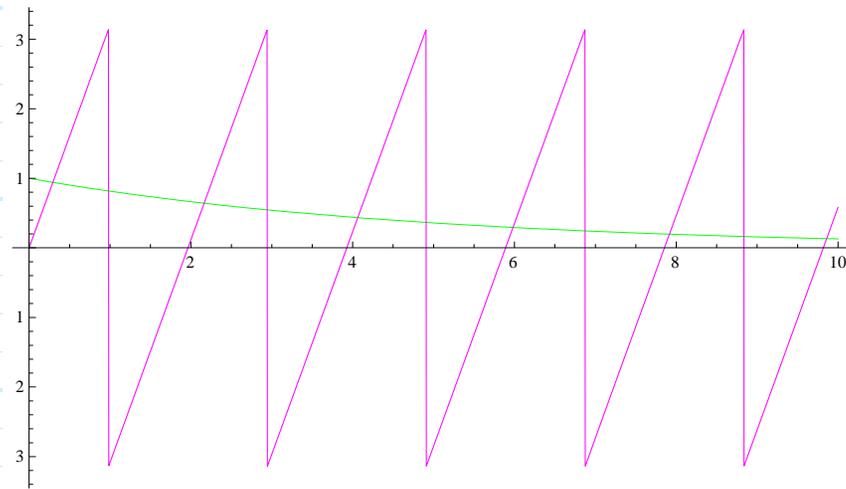
These two real-valued functions could be its **real** and **imaginary** components:

$$\begin{aligned} e^{st} &= e^{\sigma t} e^{+j\omega t} \\ &= e^{\sigma t} (\cos \omega t + j \sin \omega t) \\ &= e^{\sigma t} \cos \omega t + j e^{\sigma t} \sin \omega t \end{aligned}$$



Magnitude and phase

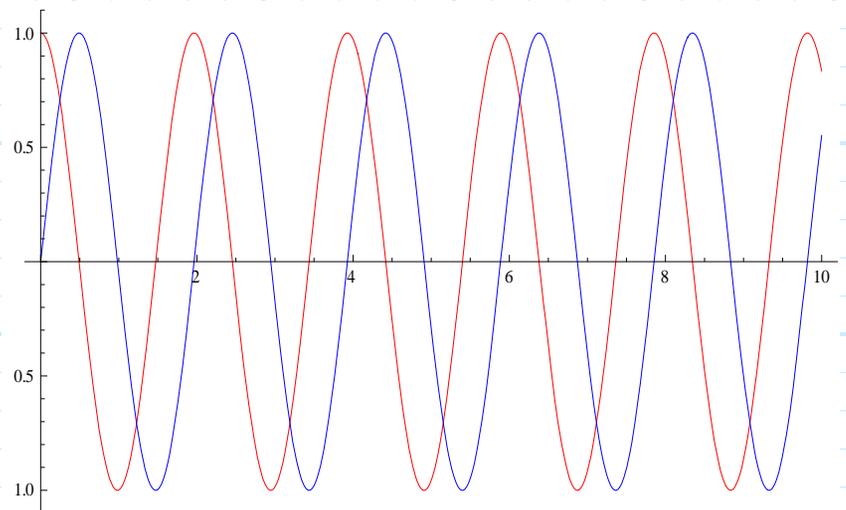
Or, the two real-valued functions could alternatively be the complex values **magnitude** and **phase**:



$$\sigma = -0.2$$

$$\omega = 3.2$$

If $\sigma = 0$, then $e^{st} = e^{+j\omega t}$, and we're back to the time-harmonic Eigen function:



$$\sigma = 0.0$$

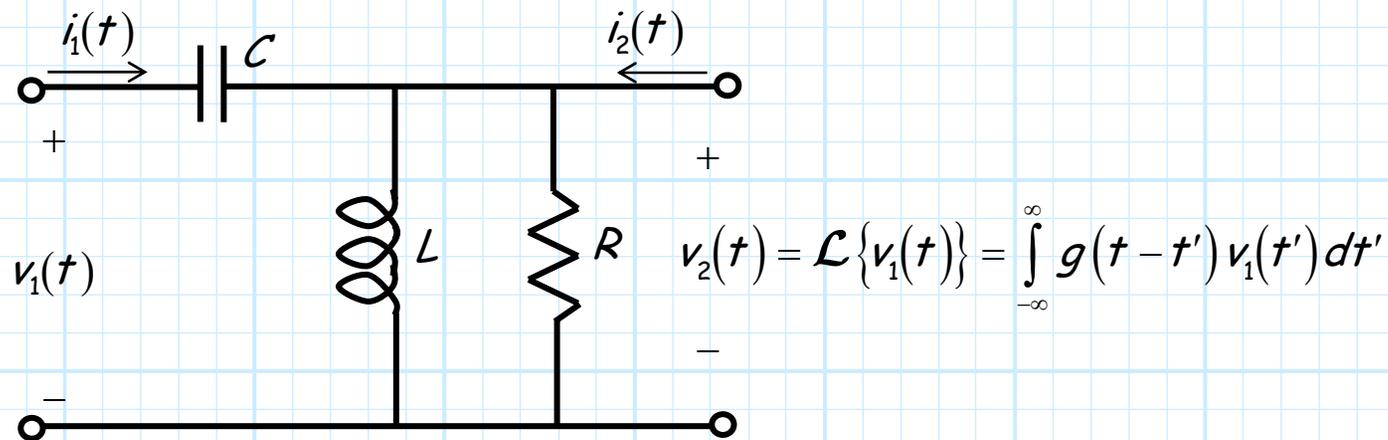
$$\omega = 3.2$$

Can we use this as a basis?

Q: What about *basis functions*? Can we use these Eigen function to *expand* a signal?

A: Sure! Instead of the Fourier Transform, the result of expanding a signal with basis function e^{st} is the **Laplace Transform**.

For example, **again** consider the following linear circuit:



$$v_2(t) = \mathcal{L}\{v_1(t)\} = \int_{-\infty}^{\infty} g(t-t')v_1(t')dt'$$

A summary

Using the Laplace transform, we can determine the **output** voltage $v_2(t)$ by:

1. Expand the **input** signal $v_1(t)$ using the basis function e^{st} :

$$V_1(s) = \int_0^{+\infty} v_1(t) e^{-st} dt \quad (\text{or use a look-up table!})$$

2. Determine the **Eigen value** of the linear operator relating $v_1(t)$ to $v_2(t)$:

$$v_2(t) = \mathcal{L}\{v_1(t)\} = \int_{-\infty}^{\infty} g(t-t') v_1(t') dt'$$

$$\Rightarrow V_2(s) = G(s) V_1(s)$$

where:

$$G(s) = \int_{-\infty}^{+\infty} g(t) e^{-st} dt$$

3. Determine $v_2(t)$ from the **inverse** Laplace transform of $V_2(s)$ (**definitely** use a look-up table!).

The Eigen values of circuit elements

Q: But how do we determine $G(s)$?

A: It's just pretty darn simple!

Again, we determine the Eigen value of each linear operator of our **three** linear circuit elements—only this time we use the Eigen function e^{st} !

$$i_R(t) = \mathcal{L}_y^R[v_R(t)] = \frac{v_R(t)}{R}$$

$$\mathcal{L}_y^R[e^{st}] = \frac{e^{st}}{R}$$

$$I_R(s) = \frac{V_R(s)}{R}$$

$$i_C(t) = \mathcal{L}_y^C[v_C(t)] = C \frac{dv_C(t)}{dt}$$

$$\mathcal{L}_y^C[e^{st}] = C \frac{d e^{st}}{dt} = sC e^{st}$$

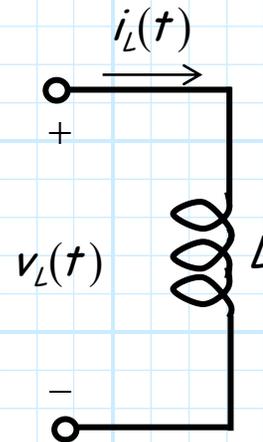
$$I_C(s) = sC V_C(s)$$

Just apply your circuits knowledge!

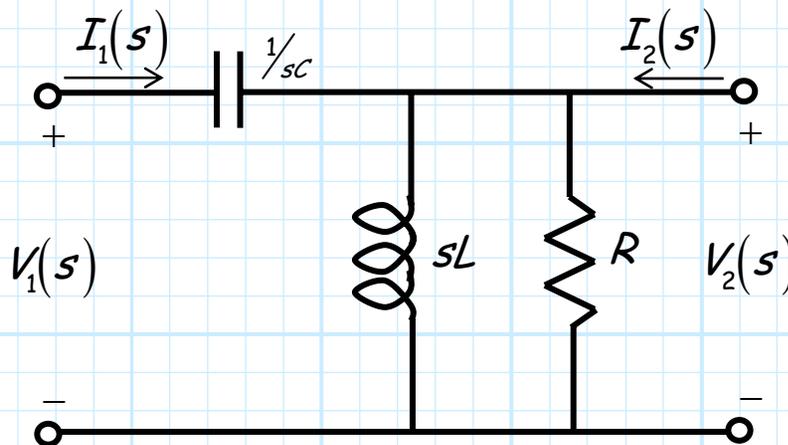
$$\mathcal{L}_y^L[v_L(t)] = i_L(t) = \frac{1}{L} \int_{-\infty}^t v_L(t') dt'$$

$$\mathcal{L}_y^L[e^{st}] = \frac{1}{L} \int_{-\infty}^t e^{st'} dt' = \frac{1}{sL} e^{st}$$

$$I_L(s) = \frac{V_L(s)}{sL}$$



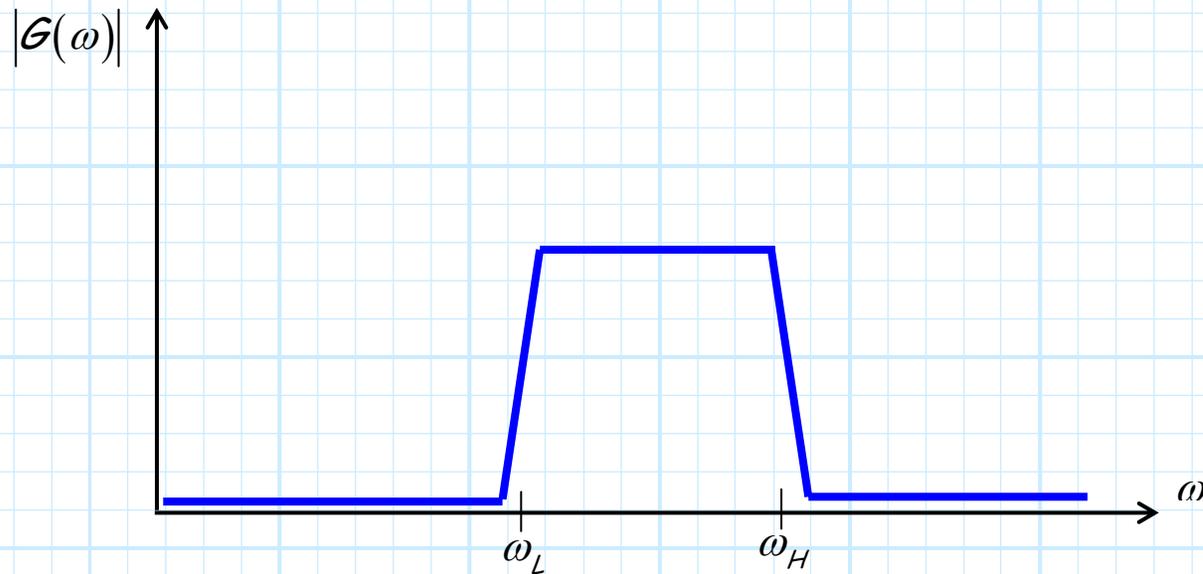
As a result we can determine the Eigen value $G(s)$ of a linear circuit by applying our **circuit theory**:



$$\frac{V_2(s)}{V_1(s)} = G_{21}(s) = \frac{sL \parallel R}{\frac{1}{sC} + sL \parallel R}$$

Frequency Bands

The Eigen value $G(\omega)$ of a linear operator is of course dependent on **frequency** ω —the **numeric value** of $G(\omega)$ depends on the frequency ω of the basis function $e^{j\omega t}$.



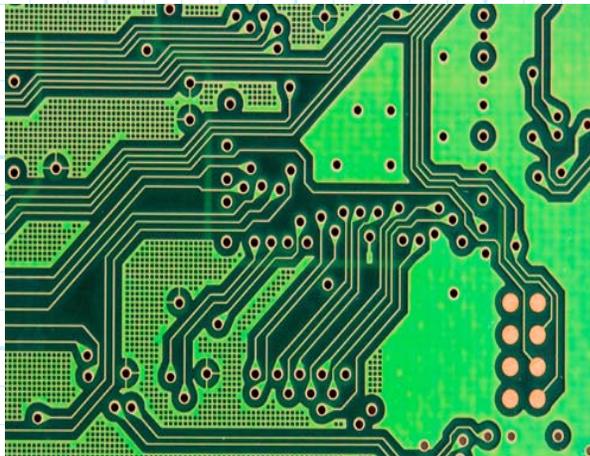
Frequency Response

The frequency ω has units of **radians/second**; it can likewise be expressed as:

$$\omega = 2\pi f$$

where f is the sinusoidal frequency in cycles/second (i.e., **Hertz**).

As a result, the function $G(\omega)$ is also known as the **frequency response** of a linear operator (e.g. a linear circuit).



The numeric value of the signal frequency f has significant **practical** ramifications to us electrical engineers, beyond that of simply determining the numeric value $G(\omega)$.

These practical ramifications include the **packaging, manufacturing, and interconnection** of electrical and electronic devices.

The problem is that every real circuit is **awash** in inductance and capacitance!

Those darn parasitics!

Q: *If this is such a problem, shouldn't we just **avoid** using capacitors and inductors?*

A: Well, capacitors and inductors are **particular useful** to us EE's.

But, even **without** capacitors and inductors, we find that our circuits are **still** awash in capacitance and inductance!

Q: ???

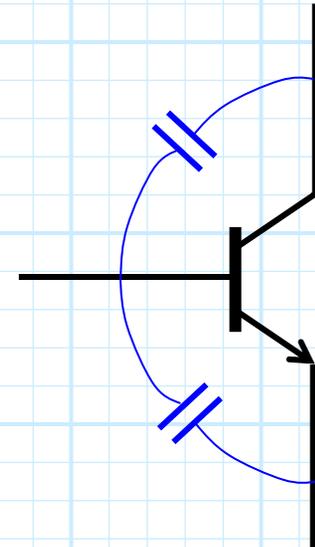
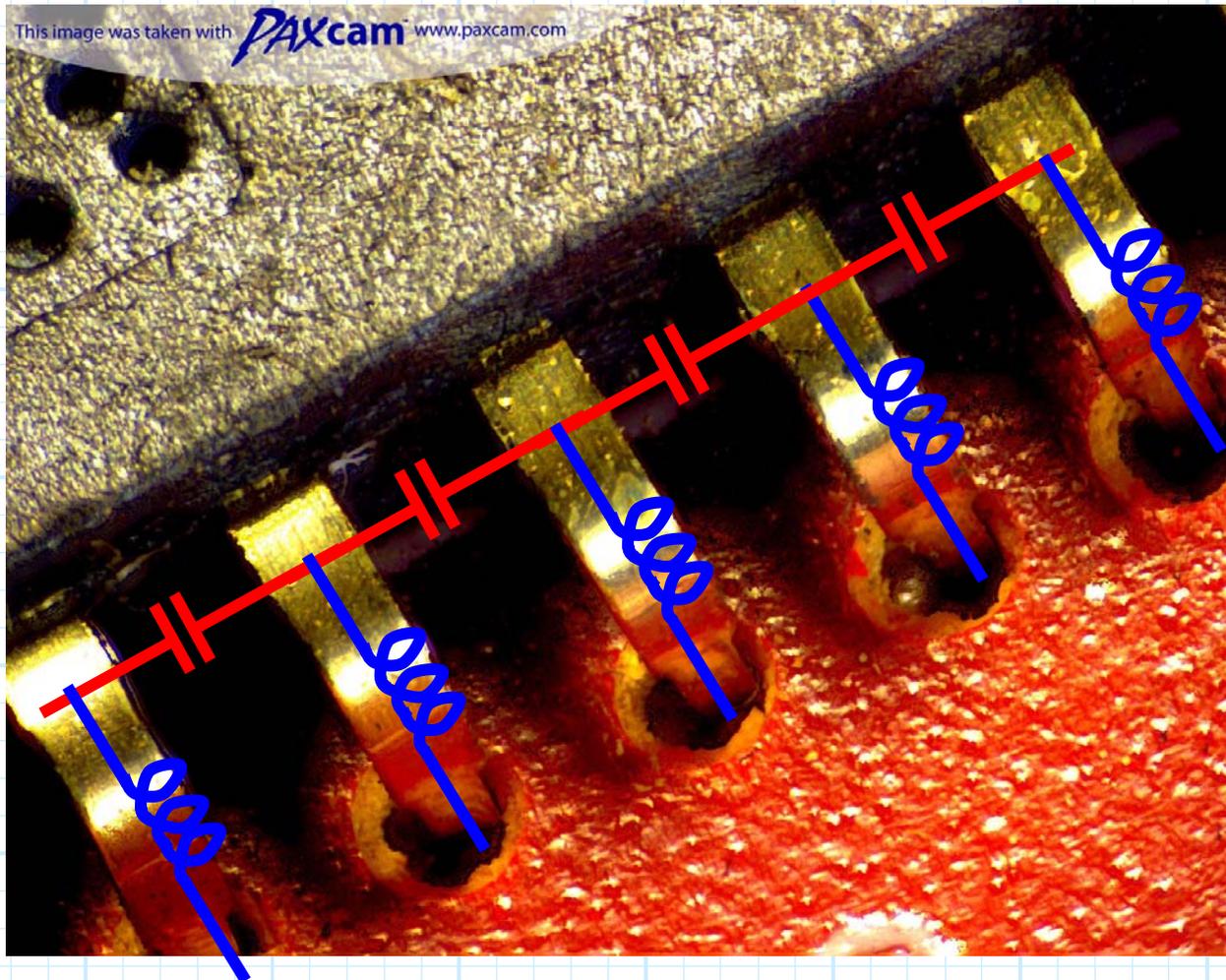
A: Every circuit that we construct will have a inherent set of **parasitic** inductance and capacitance.

Parasitic inductance and capacitance is associated with elements **other** than capacitors and inductors!



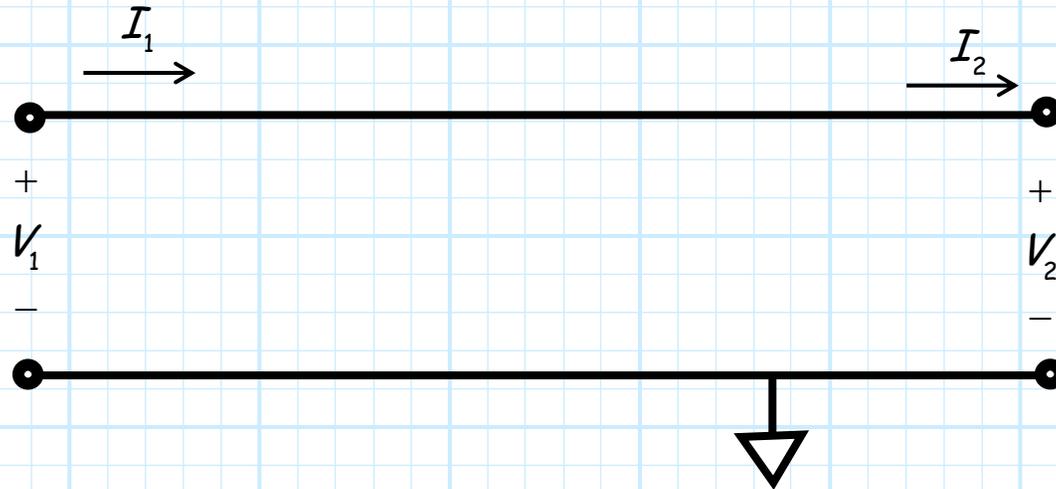
Every wire an inductor

For example, every **wire** and lead has a small inductance associated with it:



Seems simple enough...

Consider then a "wire" above a ground plane:



From KVL and KCL, we "know" that:

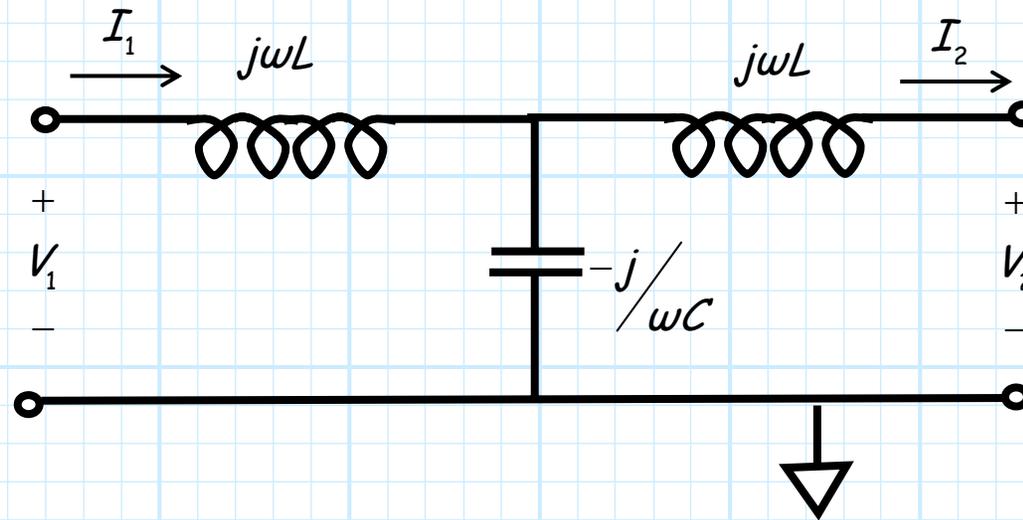
$$V_1 = V_2 \quad I_1 = I_2$$

Thus, the linear operator (for example) relating voltage V_1 to voltage V_2 has an **Eigen value equal to 1.0 for all frequencies:**

$$\frac{V_2}{V_1} = G(\omega) = 1.0$$

...but its harder than you thought!

But, the unfortunate reality is that the "wire" exhibits **inductance**, and likewise a **capacitance** between it and the ground plane



We now see that in fact the currents and voltage must be **dissimilar**:

$$V_1 \neq V_2 \quad I_1 \neq I_2$$

And so the Eigen value of the linear operator is **not** equal to 1.0!

$$\frac{V_2}{V_1} = G(\omega) \neq 1.0$$

The parasitics are small

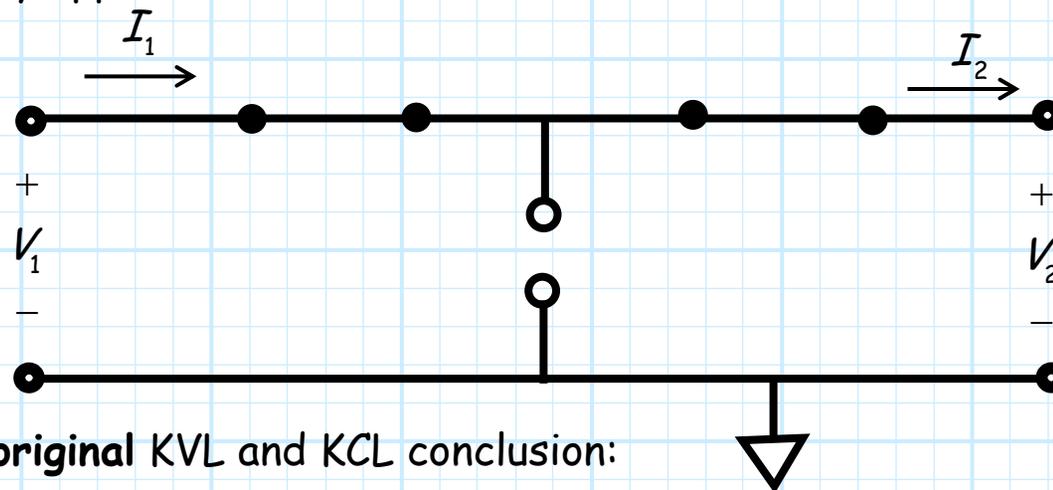
Now, these parasitic values of L and C are likely to be **very small**, so that if the frequency is "low" the **inductive impedance** is quite **small**:

$$|j\omega L| \ll 1 \quad (\text{almost a short circuit!})$$

And, the **capacitive impedance** (if the frequency is low) is quite **large**:

$$\left| \frac{-j}{\omega C} \right| \gg 1 \quad (\text{almost an open circuit!})$$

Thus, a low-frequency approximation of our wire is thus:



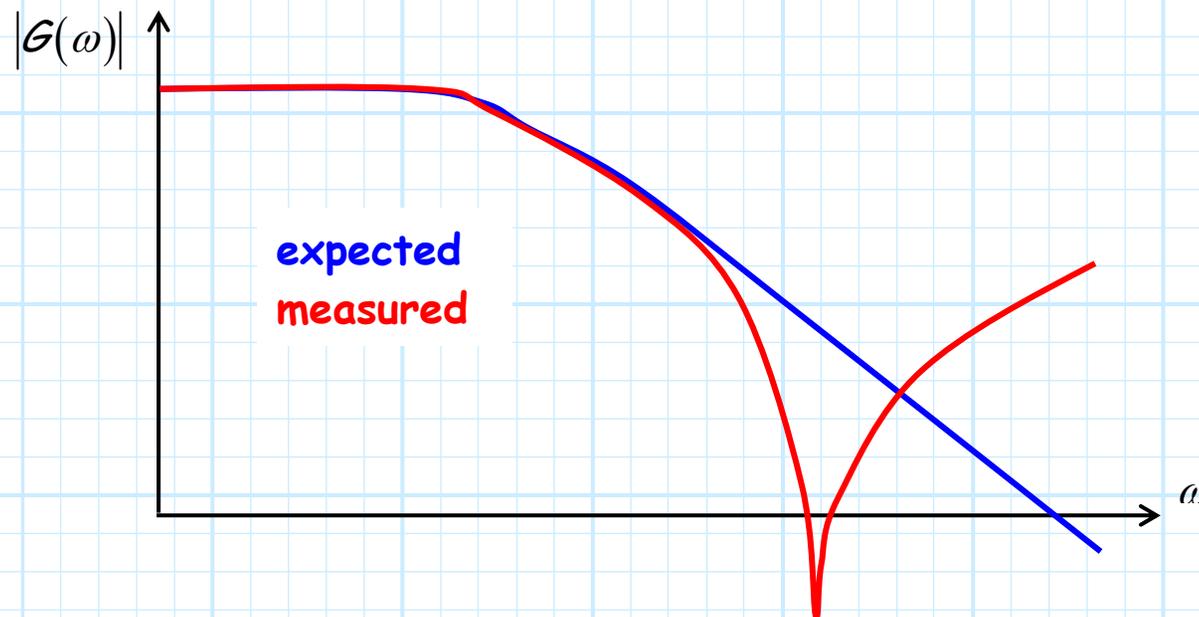
Which leads to our **original KVL and KCL** conclusion:

$$V_1 = V_2 \quad I_1 = I_2$$

Parasitics are a problem at "high" frequencies

Thus, as our signal frequency increases, the we often find that the "frequency response" $G(\omega)$ will in reality be **different** from that **predicted** by our circuit model—**unless** explicit parasitics are considered in that model.

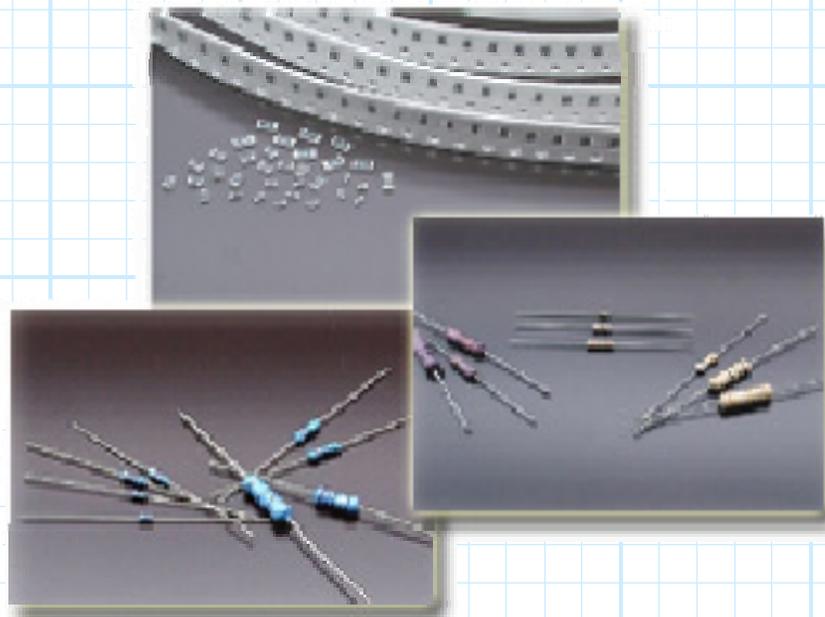
As a result, the response $G(\omega)$ may **vary** from our **expectations** as the signal frequency increases!



Frequency Bands

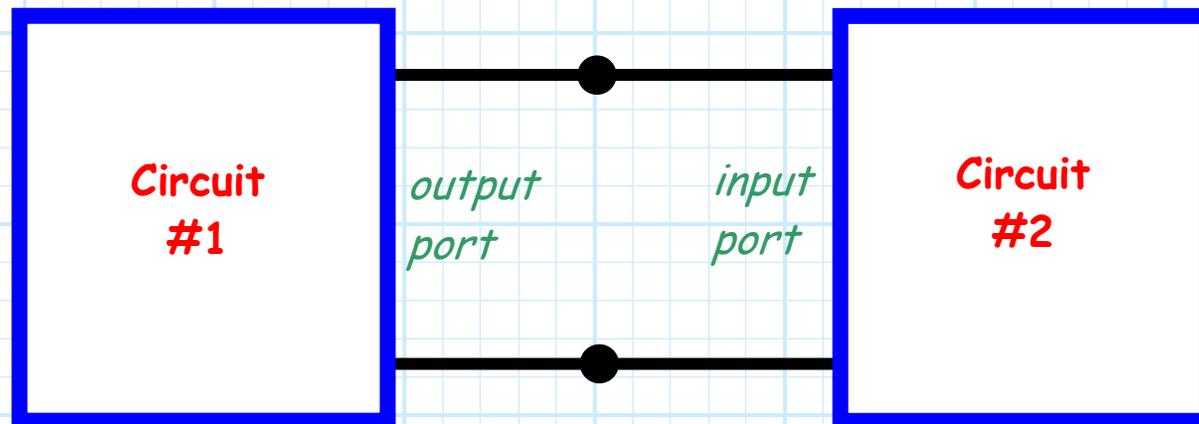
For frequencies in the kilohertz (**audio** band) or megahertz (**video** band), parasitics are generally not a problem.

However, as we move into the 100's of megahertz, or gigahertz (**RF** and **microwave** bands), the effects of parasitic inductance and capacitance are not only significant—they're **unavoidable!**



Impedance and Admittance Parameters

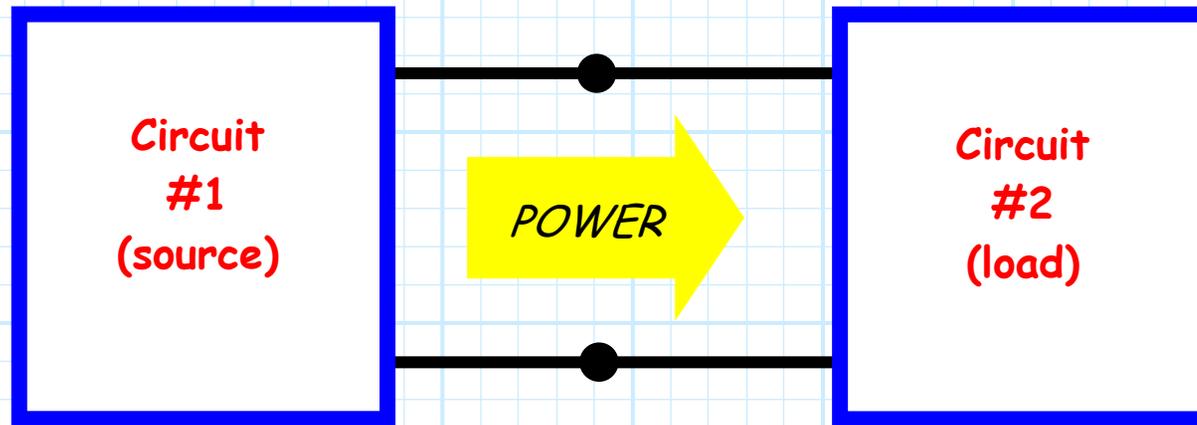
Say we wish to connect the **output** of one circuit to the **input** of another .



The terms "input" and "output" tells us that we wish for signal energy to flow **from** the output circuit **to** the input circuit.

Energy flows from source to load

In this case, the first circuit is the **source**, and the second circuit is the **load**.

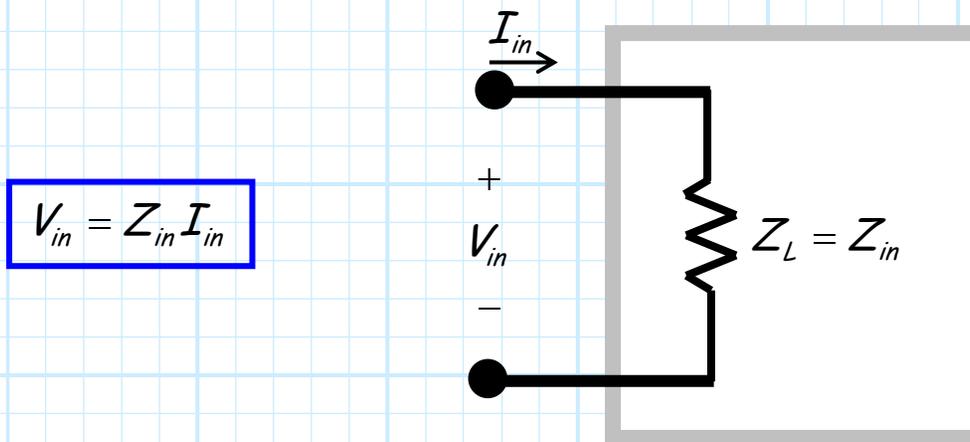
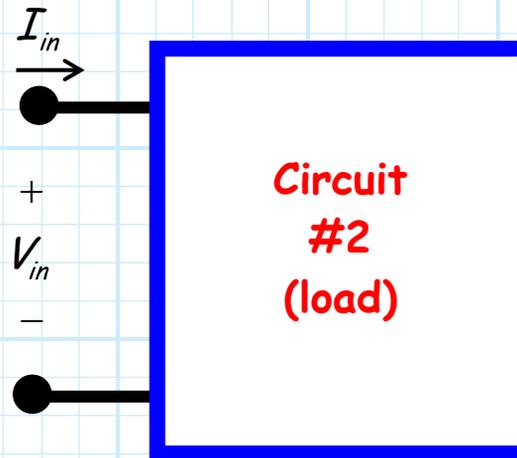


Each of these two circuits may be quite complex, but we can always simplify this problem by using **equivalent circuits**.

Load is the input impedance

For example, if we assume time-harmonic signals (i.e., eigen functions!), the load can be modeled as a simple lumped **impedance**, with a **complex** value equal to the input impedance of the circuit.

$$Z_{in} = \frac{V_{in}}{I_{in}}$$

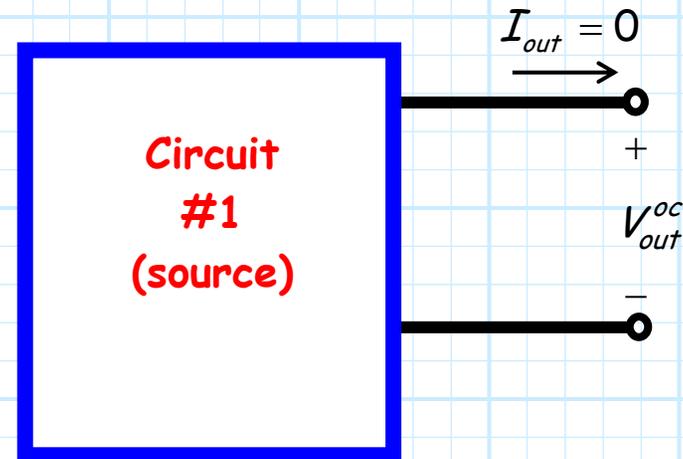


$$V_{in} = Z_{in} I_{in}$$

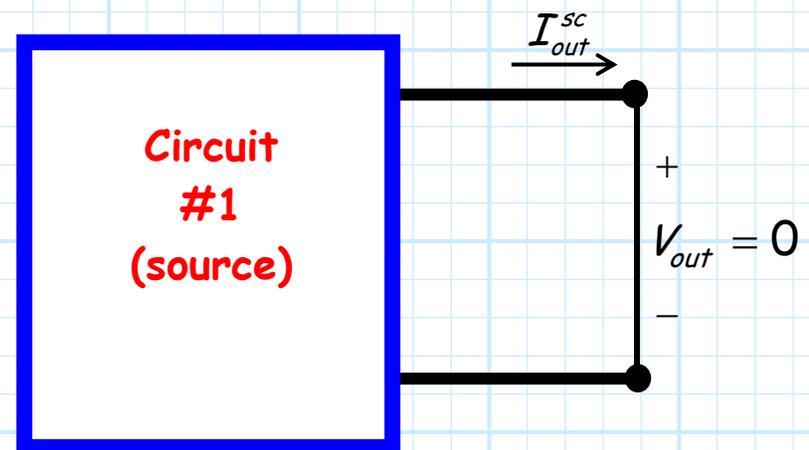
Equivalent Circuits

The source circuit can likewise be modeled using either a Thevenin's or Norton's equivalent.

This equivalent circuit can be determined by first evaluating (or measuring) the **open-circuit output voltage** V_{out}^{oc} :



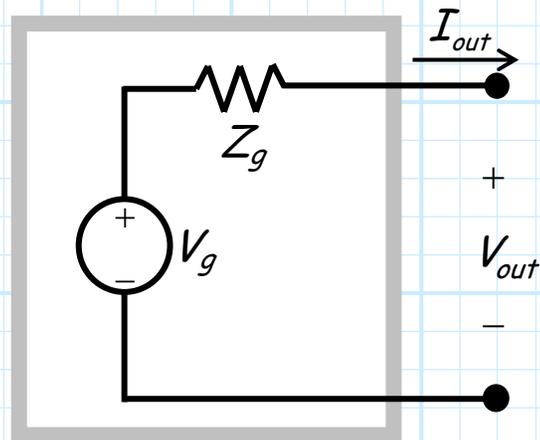
And likewise evaluating (or measuring) the **short-circuit output current** I_{out}^{sc} :



Thevenin's

From these two values (V_{out}^{oc} and I_{out}^{sc}) we can determine the Thevenin's equivalent source:

$$V_g = V_{out}^{oc} \quad Z_g = \frac{V_{out}^{oc}}{I_{out}^{sc}}$$



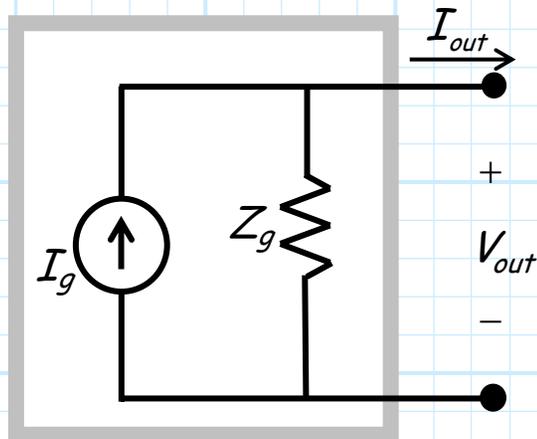
$$V_{out} = V_g - Z_g I_{out}$$

$$I_{out} = \frac{V_g - V_{out}}{Z_g}$$

Norton's

Or, we could use a Norton's equivalent circuit:

$$I_g = I_{out}^{sc} \quad Z_g = \frac{V_{out}^{oc}}{I_{out}^{sc}}$$

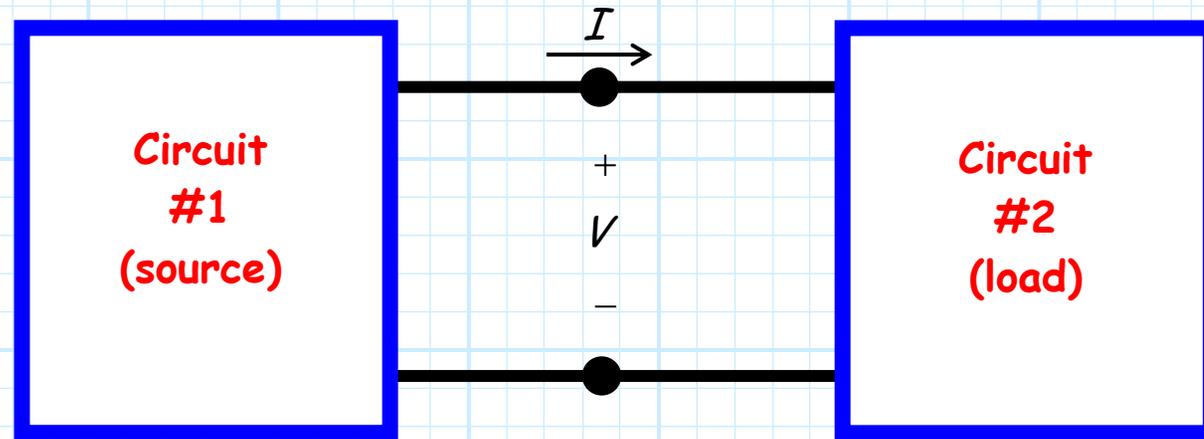


$$I_{out} = I_g - V_{out} / Z_g$$

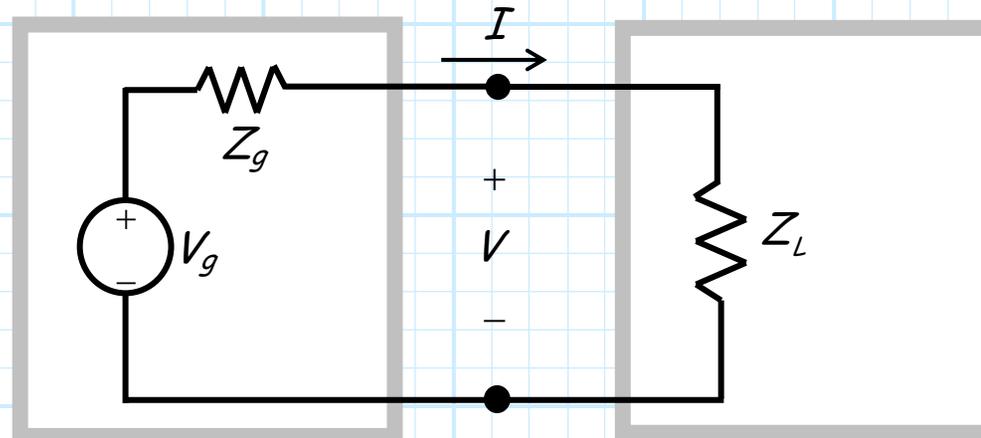
$$V_{out} = (I_g - I_{out}) Z_g$$

Circuit Model

Thus, the
entire circuit:



Can be modeled
with equivalent
circuits as:



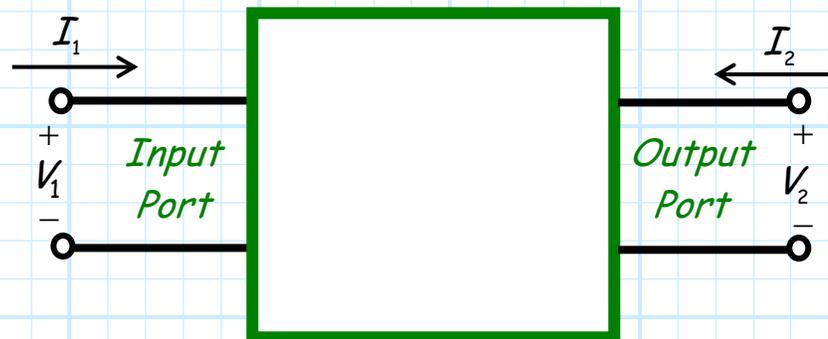
Please note again that we have assumed a **time harmonic** source, such that all the values in the circuit above (V_g , Z_g , I , V , Z_L) are **complex** (i.e., they have a **magnitude** and **phase**).

Two-Port circuits

Q: *But, circuits like filters and amplifiers are two-port devices, they have both an input and an output. How do we characterize a two-port device?*

A: Indeed, many important components are **two-port** circuits.

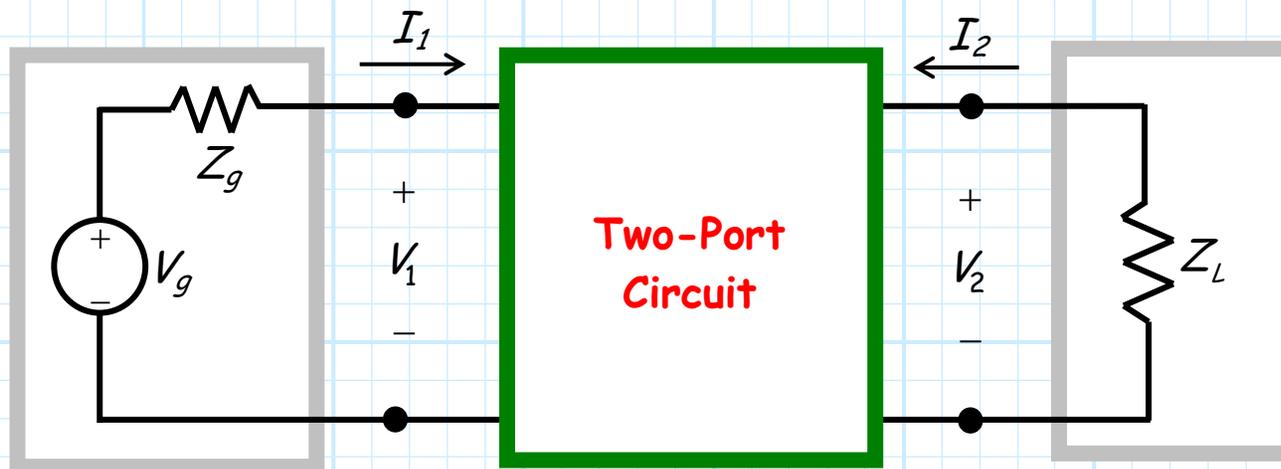
For these devices, the signal power **enters** one port (i.e., the input) and **exits** the other (the output).



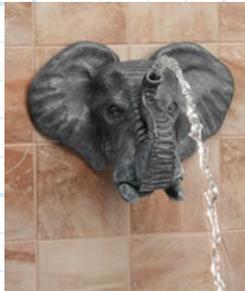
Between source and load

These two-port circuits typically do something to **alter** the signal as it passes from input to output (e.g., filters it, amplifies it, attenuates it).

We can thus assume that a **source** is connected to the **input** port, and that a **load** is connected to the **output** port.



How to characterize?



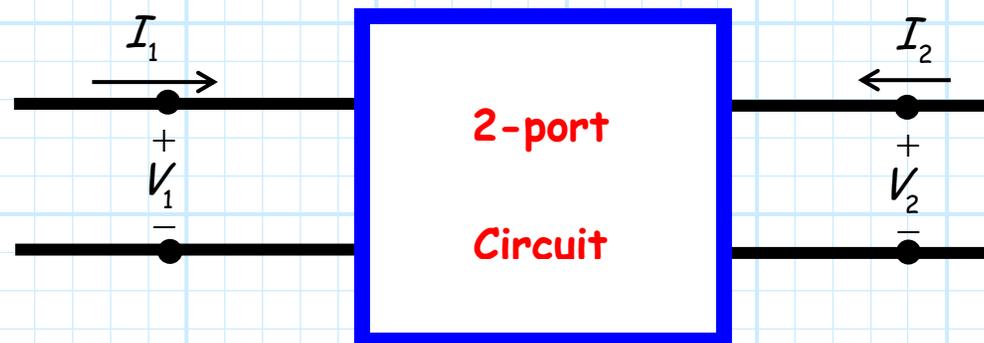
Again, the source circuit may be **quite complex**, consisting of many components. However, at least one of these components must be a **source** of energy.

Likewise, the load circuit might be **quite complex**, consisting of many components. However, at least one of these components must be a **sink** of energy.



Q: *But what about the **two-port circuit** in the middle? How do we characterize it?*

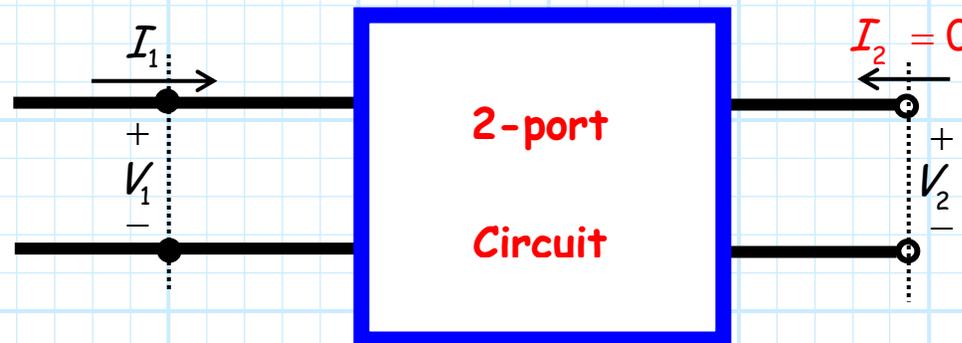
A: A linear two-port circuit is fully characterized by just four **impedance parameters!**



Do this little experiment

Note that inside the "blue box" there could be anything from a very **simple** linear circuit to a very large and **complex** linear system.

Now, say there exists a non-zero current at input **port 1** (i.e., $I_1 \neq 0$), while the current at **port 2** is known to be **zero** (i.e., $I_2 = 0$).



Say we measure/determine the **current** at port 1 (i.e., determine I_1), and we then measure/determine the **voltage** at the port 2 plane (i.e., determine V_2).

Impedance parameters

The complex ratio between V_2 and I_1 is known as the **trans-impedance parameter** Z_{21} :

$$Z_{21}(\omega) = \frac{V_2(\omega)}{I_1(\omega)}$$

Note this trans-impedance parameter is the Eigen value of the linear operator relating current $i_1(t)$ to voltage $v_2(t)$:

$$v_2(t) = \mathcal{L}\{i_1(t)\} \quad \rightarrow \quad V_2(\omega) = G_{21}(\omega)I_1(\omega)$$

Thus:

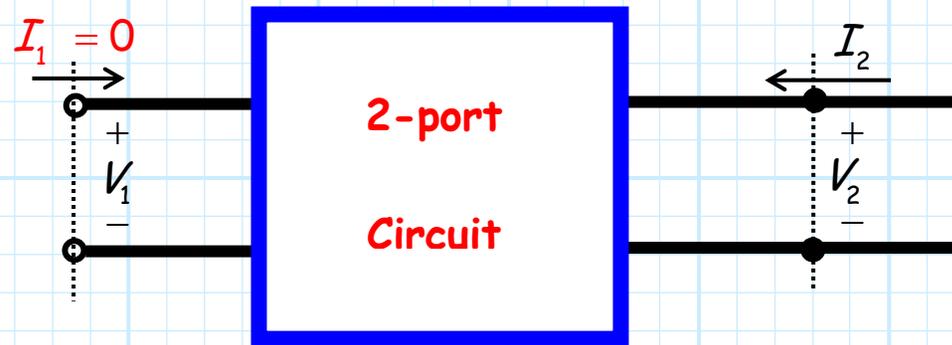
$$G_{21}(\omega) = Z_{21}(\omega)$$

Likewise, the complex ratio between V_1 and I_1 is the **trans-impedance parameter** Z_{11} :

$$Z_{11}(\omega) = \frac{V_1(\omega)}{I_1(\omega)}$$

A second experiment

Now consider the opposite situation, where there exists a non-zero current at **port 2** (i.e., $I_2 \neq 0$), while the current at port 1 is known to be **zero** (i.e., $I_1 = 0$).



The result is two more impedance parameters:

$$Z_{12}(\omega) = \frac{V_1(\omega)}{I_2(\omega)} \qquad Z_{22}(\omega) = \frac{V_2(\omega)}{I_2(\omega)}$$

Thus, more **generally**, the ratio of the current into port n and the voltage at port m is:

$$Z_{mn} = \frac{V_m}{I_n} \quad (\text{given that } I_k = 0 \text{ for } k \neq n)$$

Open circuits enforce $I=0$

Q: *But how do we ensure that one port current is zero?*



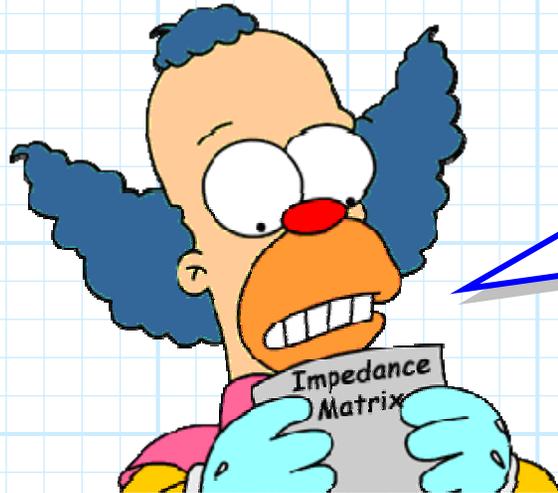
A: Place an **open** circuit **at** that port!

Placing an **open** at a port (and it must be **at** the port!) **enforces** the condition that $I = 0$.

Now, we can thus **equivalently** state the definition of trans-impedance as:

$$Z_{mn} = \frac{V_m}{I_n} \quad (\text{given that port } k \neq n \text{ is open - circuited})$$

What's the point?



*Q: As impossible as it sounds, this handout is even more **pointless** than all your previous efforts. **Why** are we studying this? After all, what is the likelihood that a device will have an **open** circuit on one of its ports?!*

A: OK, say that **neither** port is **open-circuited**, such that we have currents **simultaneously** on **both** of the two ports of our device.

Since the device is **linear**, the voltage at **one** port is due to **both** port currents.

This voltage is simply the coherent **sum** of the voltage at that port due to **each** of the two currents!

Specifically, the voltage at each port can be:

$$V_1 = Z_{11} I_1 + Z_{12} I_2$$

$$V_2 = Z_{21} I_1 + Z_{22} I_2$$

They're a function of frequency!

Thus, these four impedance parameters **completely characterizes** a linear, 2-port device.

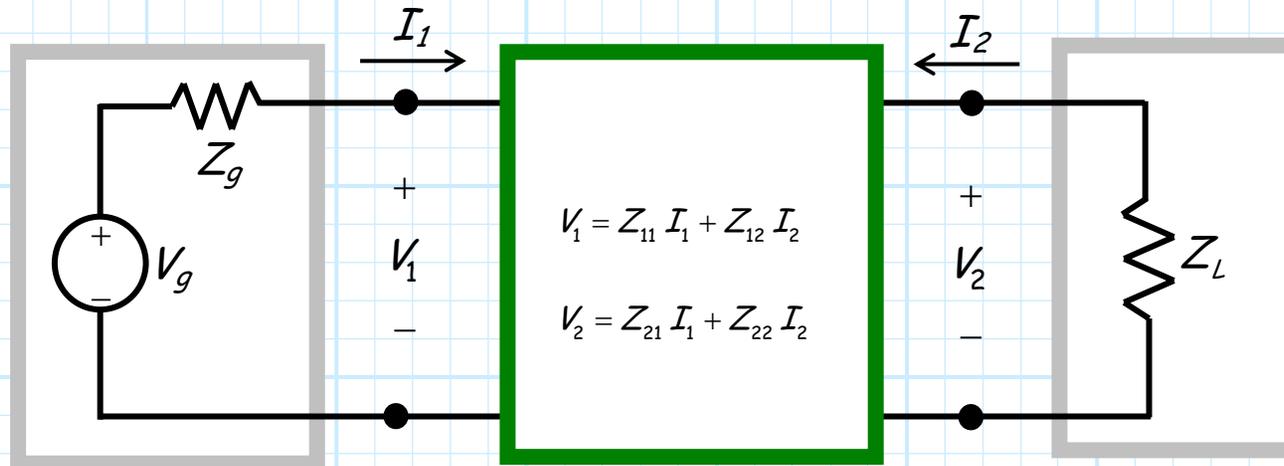
Effectively, these impedance parameters describes a 2-port device the way that Z_L describes a single-port device (e.g., a load)!



But **beware!** The values of the impedance matrix for a particular device or circuit, just like Z_L , are **frequency dependent!**

A complete equivalent circuit

Now, we can use our equivalent circuits to model this system:



Note in this circuit there are **4 unknown values**—two voltages (V_1 and V_2), and two currents (I_1 and I_2).

→ Our job is to **determine** these 4 unknown values!

Let's do some algebra!

Let's begin by looking at the source, we can determine from KVL that:

$$V_g - Z_g I_1 = V_1$$

And so with a bit of algebra:

$$I_1 = \frac{V_g - V_1}{Z_g} \quad (\leftarrow \text{look, Ohm's Law!})$$

Now let's look at our two-port circuit. If we know the impedance matrix (i.e., all **four trans-impedance** parameters), then:

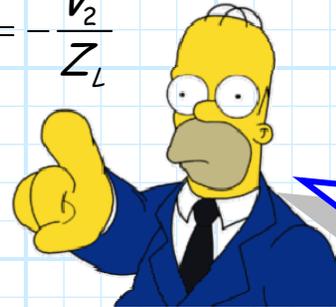
$$V_1 = Z_{11} I_1 + Z_{12} I_2$$

$$V_2 = Z_{21} I_1 + Z_{22} I_2$$

Watch the minus sign!

Finally, for the load:

$$I_2 = -\frac{V_2}{Z_L}$$



Q: *Are you sure this is correct? I don't recall there being a **minus sign** in Ohm's Law.*

A: Be very careful with the notation.

Current I_2 is defined as positive when it is flowing into the two port circuit. This is the notation required for the impedance matrix.

Thus, positive current I_2 is flowing out of the load impedance—the opposite convention to Ohm's Law.

This is why the **minus sign** is required.

A very good thing

Now let's **take stock** of our results. Notice that we have compiled **four** independent equations, involving our **four** unknown values:

$$I_1 = \frac{V_g - V_1}{Z_g}$$

$$I_2 = -\frac{V_2}{Z_L}$$

$$V_1 = Z_{11} I_1 + Z_{12} I_2$$

$$V_2 = Z_{21} I_1 + Z_{22} I_2$$

Q: *Four equations and four unknowns! That sounds like a very good thing!*



A: It is! We can apply a bit of **algebra** and solve for the unknown currents and voltages:

$$I_1 = V_g \frac{Z_{22} + Z_L}{(Z_{11} + Z_g)(Z_{22} + Z_L) - Z_{12}Z_{21}}$$

$$I_2 = -V_g \frac{Z_{21}}{(Z_{11} + Z_g)(Z_{22} + Z_L) - Z_{12}Z_{21}}$$

$$V_1 = V_g \frac{Z_{11}(Z_{22} + Z_L) - Z_{12}Z_{21}}{(Z_{11} + Z_g)(Z_{22} + Z_L) - Z_{12}Z_{21}}$$

$$V_2 = V_g \frac{Z_L Z_{21}}{(Z_{11} + Z_g)(Z_{22} + Z_L) - Z_{12}Z_{21}}$$

Admittance Parameters

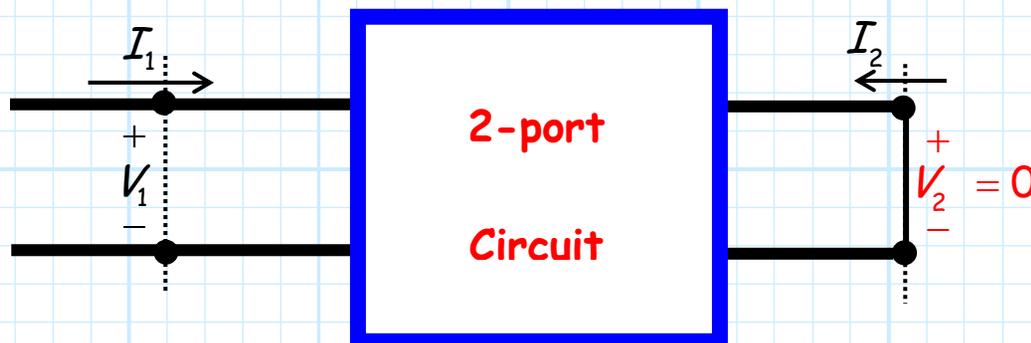
Q: Are impedance parameters the *only* way to characterize a 2-port linear circuit?

A: Hardly! Another method uses **admittance parameters**.

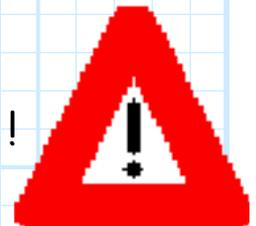
The elements of the Admittance Matrix are the **trans-admittance** parameters Y_{mn} , defined as:

$$Y_{mn} = \frac{I_m}{V_n} \quad (\text{given that } V_k = 0 \text{ for } k \neq n)$$

Note here that the **voltage** at one port **must** be equal to **zero**. We can ensure that by simply placing a **short circuit** at the zero-voltage port!



Note that $Y_{mn} \neq 1/Z_{mn}$!



Short circuits enforce $V=0$

Now, we can **equivalently** state the definition of trans-admittance as:

$$Y_{mn} = \frac{I_m}{V_n} \quad (\text{given that all ports } k \neq n \text{ are short-circuited})$$

Just as with the trans-impedance values, we can use the trans-admittance values to evaluate general circuit problems, where **none** of the ports have zero voltage.

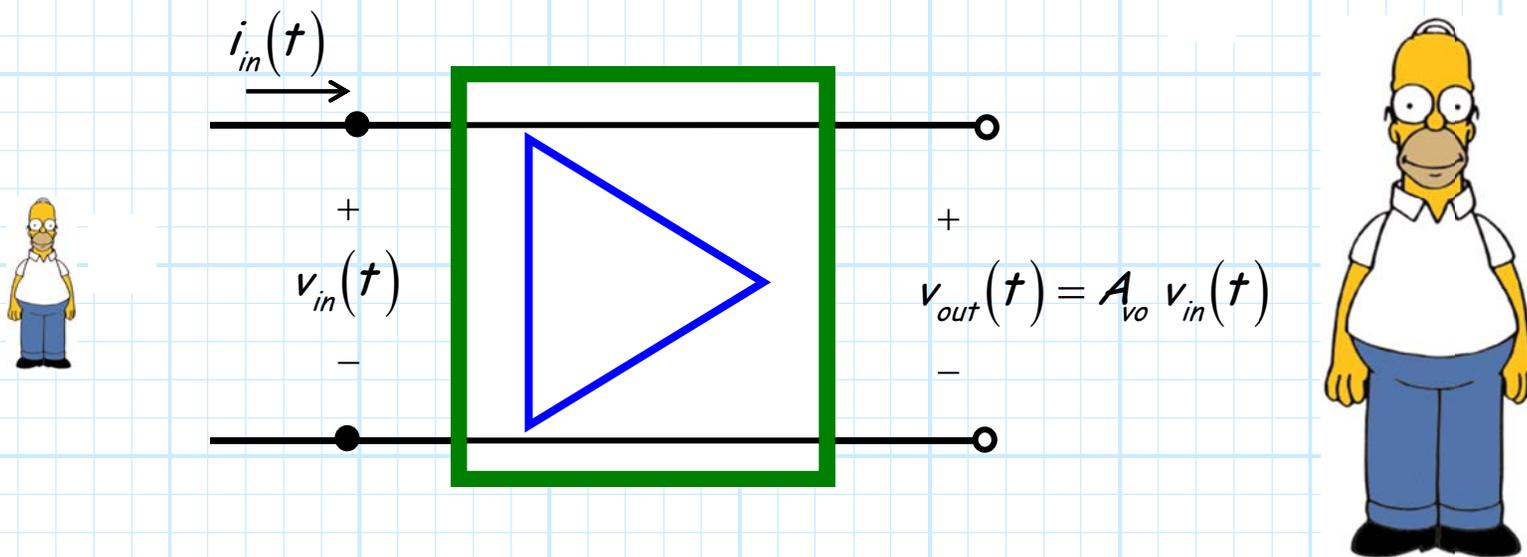
Since the device is **linear**, the current at any **one** port due to **all** the port currents is simply the coherent **sum** of the currents at that port due to **each** of the port voltages!

$$I_1 = Y_{11} V_1 + Y_{12} V_2$$

$$I_2 = Y_{21} V_1 + Y_{22} V_2$$

Amplifiers

An **ideal** amplifier is a two-port circuit that takes an input signal $v_{in}(t)$ and reproduces it **exactly** at its output, only with a **larger** magnitude!



The real value A_{vo} is the **open-circuit voltage gain** of this ideal amplifier, and has a magnitude much larger than unity ($A_{vo} \gg 1$).

We actually can find $g(t)$!

Now, let's express this result using our knowledge of **linear circuit theory**!

Recall, the output $v_{out}(t)$ of a linear device can be determined by **convolving** its input $v_{in}(t)$ with the device **impulse response** $g(t)$:

$$v_{out}(t) = \int_{-\infty}^t g(t-t') v_{in}(t') dt'$$

Q: *Yikes! What is the impulse response of this ideal amp? How can we determine it?*

A: It's actually quite **simple**!

Remember, the **impulse response** of linear circuit is just the output that results when the **input** is an impulse function $\delta(t)$.

Every function an Eigen function

Since the output of an ideal amplifier is just the input multiplied by A_{vo} , we conclude if $v_{in}(t) = \delta(t)$:

$$g(t) = v_{out}(t) = A_{vo} \delta(t)$$

Thus:

$$\begin{aligned} v_{out}(t) &= \int_{-\infty}^t g(t-t') v_{in}(t') dt' \\ &= \int_{-\infty}^t A_{vo} \delta(t-t') v_{in}(t') dt' \\ &= A_{vo} \int_{-\infty}^t \delta(t-t') v_{in}(t') dt' \\ &= A_{vo} v_{in}(t) \end{aligned}$$

→ **Any and every function $v_{in}(t)$ is an Eigen function of an ideal amplifier!!**

And now the Eigen value

Now, we can determine the **Eigen value** of this linear operator relating input to output:

$$v_{out}(t) = \mathcal{L}\{v_{in}(t)\}$$

Recall this Eigen value is found from the **Fourier transform** of the impulse response:

$$\begin{aligned} G(\omega) &= \int_{-\infty}^{\infty} h(t) e^{-j\omega t} dt \\ &= \int_{-\infty}^{\infty} A_{vo} \delta(t) e^{-j\omega t} dt \\ &= A_{vo} + j0 \\ &= A_{vo} e^{j0} \end{aligned}$$

This result, although simple, has an interesting interpretation...

DC to daylight

...it means that the amplifier exhibits gain of A_{vo} for sinusoidal signals of **any** and **all** frequencies!



BUT, there is one **big** problem with an ideal amplifier:

→ They are **impossible** to build!!

Real amplifier have finite bandwidths

The **ideal** amplifier has a frequency response of $|G(\omega)| = A_{vo}$.

Note this means that the amplifier gain is A_{vo} for **all** frequencies $0 < \omega < \infty$ (D.C. to daylight!).

The **bandwidth** of the **ideal** amplifier is therefore **infinite!**

- * Since every electronic device will exhibit **some** amount of inductance, capacitance, and resistance, every device will have a **finite** bandwidth.
- * In other words, there will be frequencies ω where the device does **not work!**
- * From the standpoint of an amplifier, "not working" means $|G(\omega)| \ll A_{vo}$ (i.e., **low gain**).

→ Amplifiers therefore have **finite** bandwidths.

Amplifier bandwidth

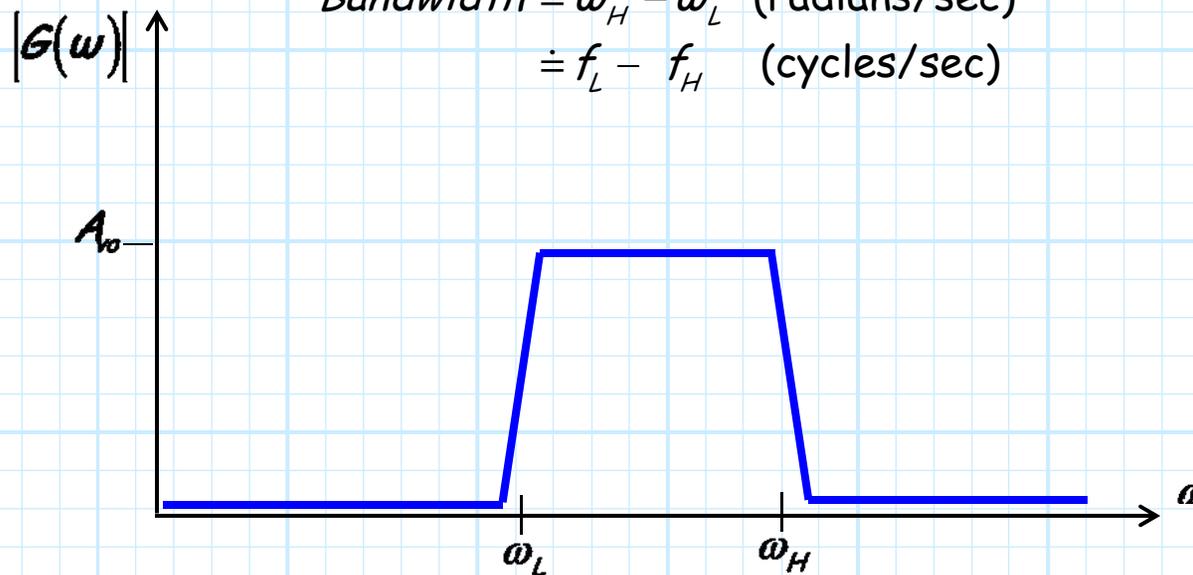
There is a range of frequencies ω between ω_L and ω_H where the gain will (approximately) be A_{vo} .

For frequencies outside this range, the gain will typically be small (i.e. $|G(\omega)| \ll A_{vo}$):

$$|G(\omega)| = \begin{cases} \approx A_{vo} & \omega_L < \omega < \omega_H \\ \ll A_{vo} & \omega < \omega_L, \omega > \omega_H \end{cases}$$

The **width** of this frequency range is called the amplifier **bandwidth**:

$$\begin{aligned} \text{Bandwidth} &\doteq \omega_H - \omega_L \quad (\text{radians/sec}) \\ &\doteq f_H - f_L \quad (\text{cycles/sec}) \end{aligned}$$



Wideband is desirable

One result of a **finite bandwidth** is that the amplifier impulse response is **not** an impulse function !

$$h(t) = \int_{-\infty}^{\infty} H(\omega) e^{+j\omega t} dt \neq A_{vo} \delta(t)$$

therefore **generally** speaking:

$$v_{out}(t) \neq A_{vo} v_{in}(t) !!$$

However, if an input signal **spectrum** $V_{in}(\omega)$ lies completely **within** the amplifier bandwidth, then we find that will (approximately) behave like an **ideal** amplifier:

$$v_{out}(t) \cong A_{vo} v_{in}(t) \quad \text{if } V_{in}(\omega) \text{ is within the amplifier bandwidth}$$

As a result, **maximizing** the bandwidth of an amplifier is a typically and important **design goal**!

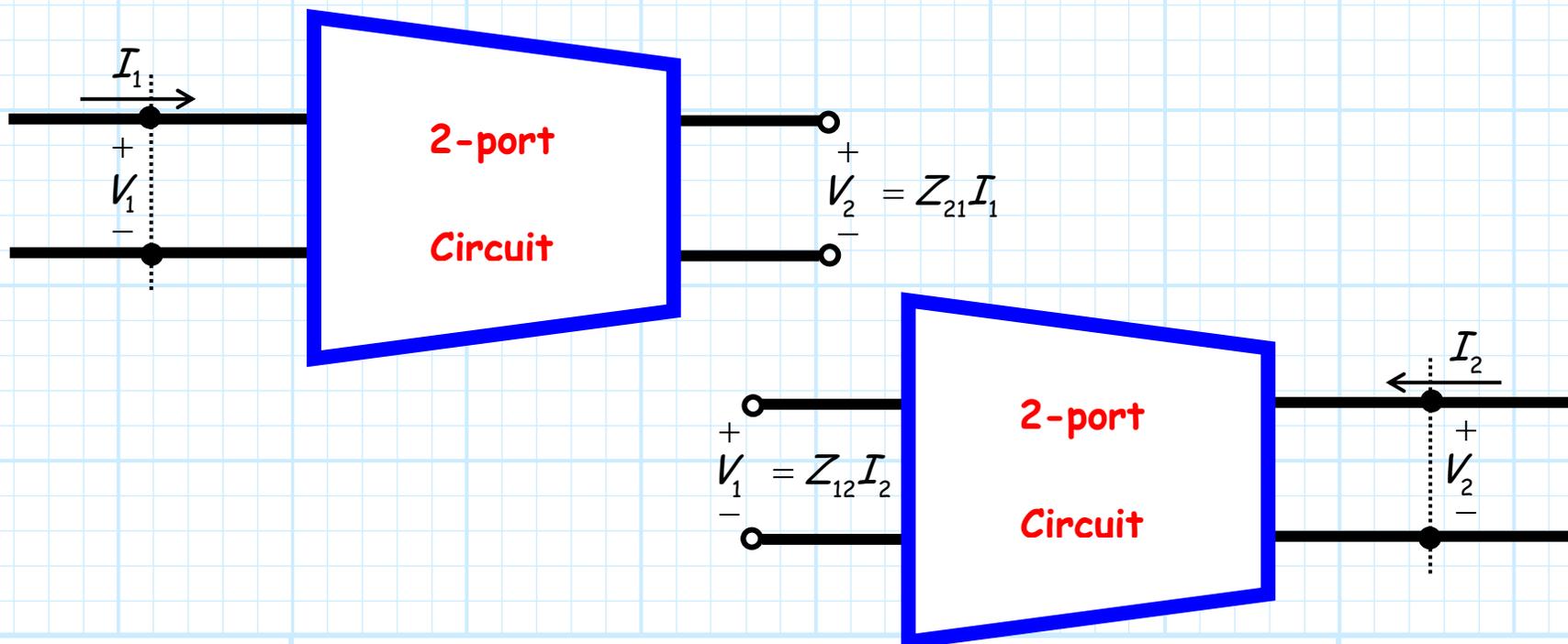
Amplifier Gain

One interesting characteristic of an amplifier is that it is a **unilateral** device—it makes a big difference **which end** you use as the **input**!

Most passive linear circuits (e.g., using only R, L and C) are **reciprocal**. With respect to a 2-port device, **reciprocity** means:

$$Z_{12}(\omega) = Z_{21}(\omega) \quad \text{and} \quad Y_{12}(\omega) = Y_{21}(\omega)$$

For example, consider these two **open-circuit voltage** measurements:



Most linear circuits are reciprocal...

If this linear two-port circuit is also **reciprocal**, then when the two currents I_1 and I_2 are equal, so too will be the resulting **open-circuit** voltages V_1 and V_2 !

Thus, a **reciprocal** 2-port circuit will have the property:

$$V_1 = V_2 \quad \text{when} \quad I_1 = I_2$$

Note this would likewise mean that:

$$\frac{V_2}{I_1} = \frac{V_1}{I_2}$$

And since (because of the **open-circuits!**):

$$V_2 = Z_{21}I_1 \quad \text{and} \quad V_1 = Z_{12}I_2$$

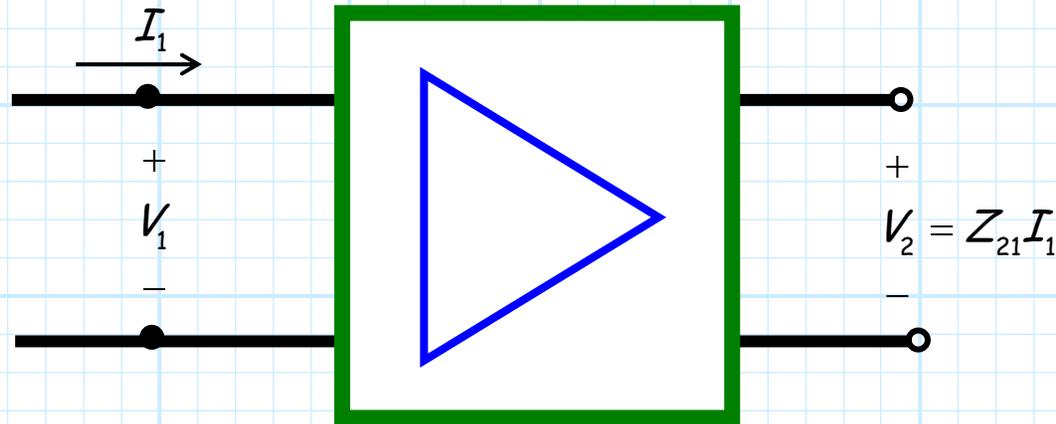
We can conclude from this "experiment" that these trans-impedance parameters of a **reciprocal** 2-port device are **equal**:

$$Z_{12}(\omega) = Z_{21}(\omega)$$

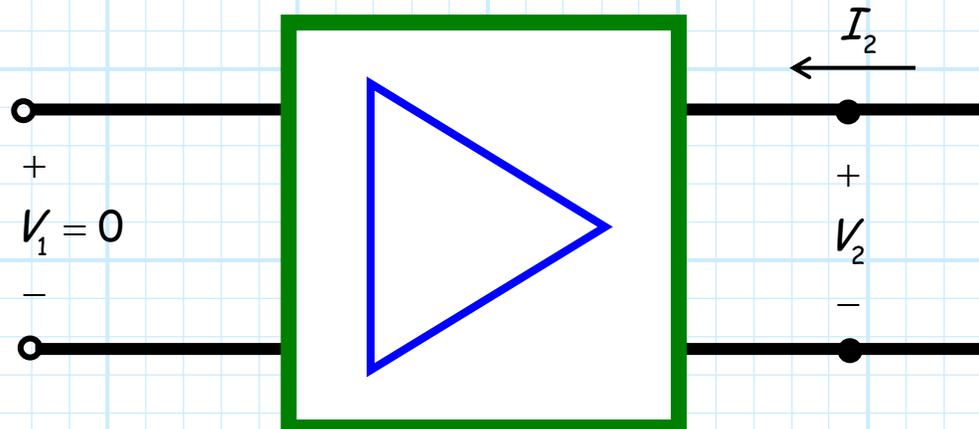
...but amplifiers are not!

Contrast this with an amplifier.

A **current** on the input port will indeed produce a **voltage** on an open-circuited output:



However, **amplifiers are not reciprocal**. Placing the same current at the **output** will **not** create the equal voltage on the input—in fact, it will produce **no voltage at all!**



Amps are unilateral: an input and output

Since for this **open-circuited** input port we know that:

$$Z_{12} = \frac{V_1}{I_2},$$

the fact that voltage produced at the input port is zero ($V_1 = 0$) means the trans-impedance parameter Z_{12} is likewise **zero** (or nearly so) for unilateral amplifiers:

$$Z_{12}(\omega) = 0 \quad (\text{for amplifiers})$$

Thus, the two equations describing an amplifier (a two-port device) **simplify nicely**.

Here's the simplification

Beginning with:

$$V_1 = Z_{11} I_1 + Z_{12} I_2$$

$$V_2 = Z_{21} I_1 + Z_{22} I_2$$

Now since $Z_{12} = 0$, we find:

$$V_1 = Z_{11} I_1$$

$$V_2 = Z_{21} I_1 + Z_{22} I_2$$

Q: *Gee; I'm sort of **unimpressed** by this simplification—I was hoping the result would be a little more—simple.*

A: Actually, the two equations above represent a **tremendous** simplification—it completely **decouples** the input port from the output, and it allows us to assign very real **physical interpretations** to the remaining impedance parameters!

To see all these benefits (**try to remain calm**), we will now make a few changes in the **notation**.

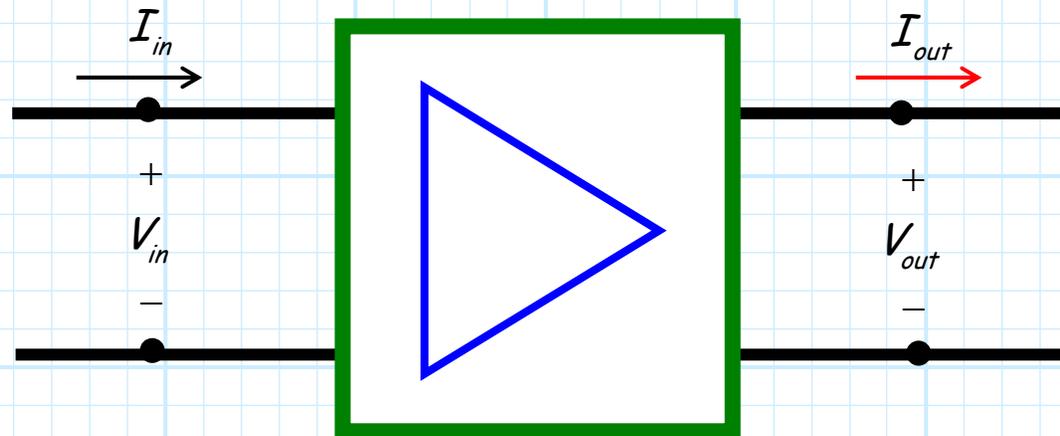
A slight change in notation

First we explicitly denote voltage V_1 as V_{in} , and likewise V_2 as V_{out} (the same with currents I).

Additionally, we change the current **definition** at the output port, **reversing** the direction of positive current as flowing **outward** from the output port. Thus:

$$I_{out} = -I_2$$

And so, a **tidy** summary:



$$V_{in} = Z_{11} I_{in}$$

$$V_{out} = Z_{21} I_{in} - Z_{22} I_{out}$$

The input is independent of the output!

From this summary, it is evident that the relationship between the **input** current and **input** voltage is determined by impedance parameter Z_{11} —and Z_{11} **only**:

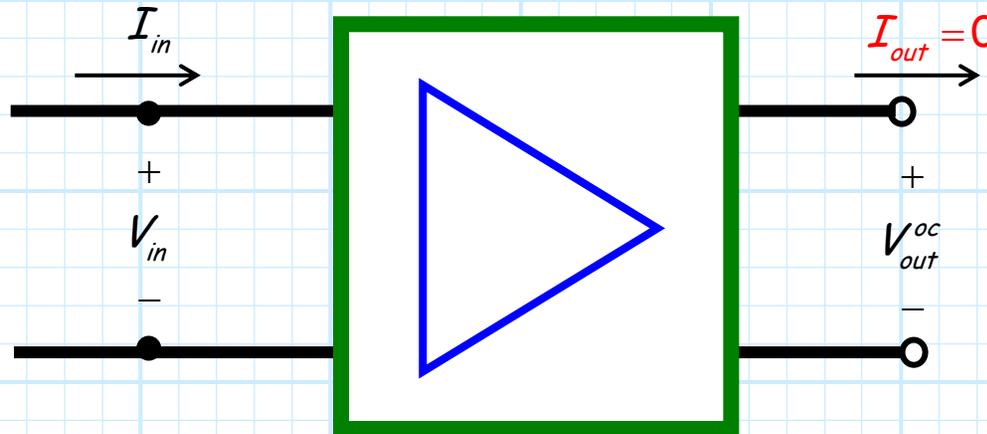
$$Z_{11} = \frac{V_{in}}{I_{in}}$$

Thus, the impedance parameter Z_{11} is known as the **input impedance** Z_{in} of an (unilateral!) amplifier:

$$Z_{in}(\omega) \doteq \frac{V_{in}(\omega)}{I_{in}(\omega)} = Z_{11}(\omega)$$

The open-circuit output voltage

Now, consider the case where the output port of the amplifier is **open-circuited** ($I_{out} = 0$):



The (**open-circuit**) output voltage is therefore simply:

$$\begin{aligned} V_{out} &= Z_{21} I_{in} - Z_{22} I_{out} \\ &= Z_{21} I_{in} - Z_{22} (0) \\ &= Z_{21} I_{in} \end{aligned}$$

The **open-circuit output voltage** is thus proportional to the **input current**.

Open-circuit trans-impedance

The proportionality constant is the impedance parameter Z_{21} —a value otherwise known as the **open-circuit trans-impedance** Z_m :

$$Z_m(\omega) \doteq \frac{V_{out}^{oc}(\omega)}{I_{in}(\omega)} = Z_{21}(\omega)$$

Thus, an (unilateral!) amplifier can be described as:

$$V_{in} = Z_{in} I_{in}$$

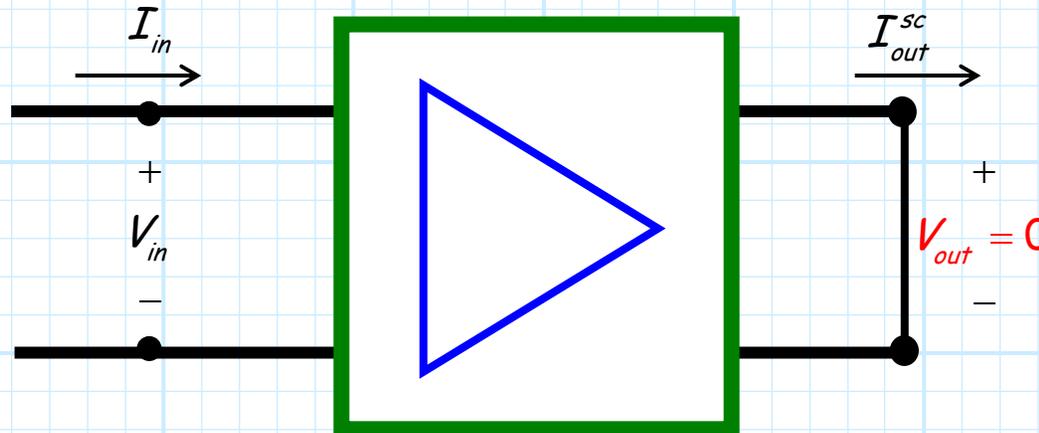
$$V_{out} = Z_m I_{in} - Z_{22} I_{out}$$

Short-circuit output current

Q: What about impedance parameter Z_{22} ; does it have any physical meaning?

A: It sure does!

Consider now the result of **short-circuiting** the amplifier output ($\therefore V_{out} = 0$):



Since $V_{out} = 0$:

$$V_{out} = 0 = Z_m I_{in} - Z_{22} I_{out}^{sc}$$

we can quickly determine the **short-circuit output current**:

$$I_{out}^{sc} = \frac{Z_m I_{in}}{Z_{22}}$$

The output impedance

Q: *I'm not seeing the significance of this result!?*

A: Let's rearrange to determine Z_{22} :

$$Z_{22} = \frac{Z_m I_{in}}{I_{out}^{sc}}$$

Note the **numerator**—it is the **open-circuit voltage** $V_{out}^{oc} = Z_m I_{in}$, and so:

$$Z_{22} = \frac{Z_m I_{in}}{I_{out}^{sc}} = \frac{V_{out}^{oc}}{I_{out}^{sc}}$$

Of course, you remember that the ratio of the **open-circuit voltage** to **short-circuit current** is the **output impedance** of a source:

$$Z_{out} \doteq \frac{V_{out}^{oc}}{I_{out}^{sc}} = Z_{22}$$

These equations look familiar!

Thus, the **output impedance** of an (unilateral) amplifier is the impedance parameter Z_{22} , and so:

$$V_{in} = Z_{in} I_{in}$$

$$V_{out} = Z_m I_{in} - Z_{out} I_{out}$$

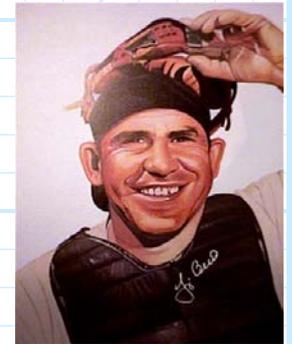
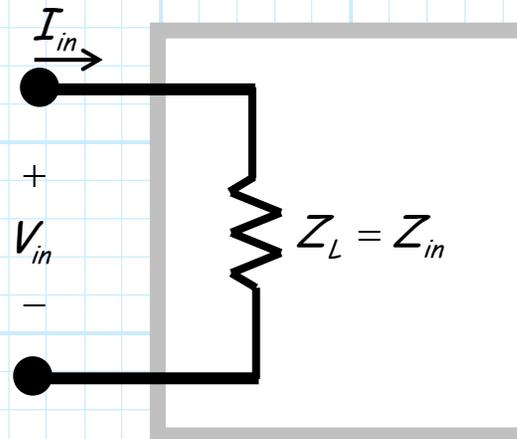
Q: *It's déjà vu all over again; haven't we seen equations like this before?*

A: Yes! Recall the first (i.e., input) equation:

$$V_{in} = Z_{in} I_{in}$$

is that of a simple **load impedance**:

$$V_{in} = Z_{in} I_{in}$$



Looks like a Thevenin's source

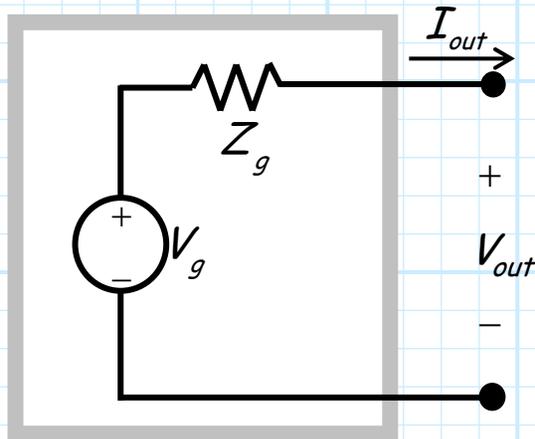
And the **second** (i.e., output) amplifier equation:

$$V_{out} = Z_m I_{in} - Z_{out} I_{out}$$

is of the form of a **Thevenin's source**:

where:

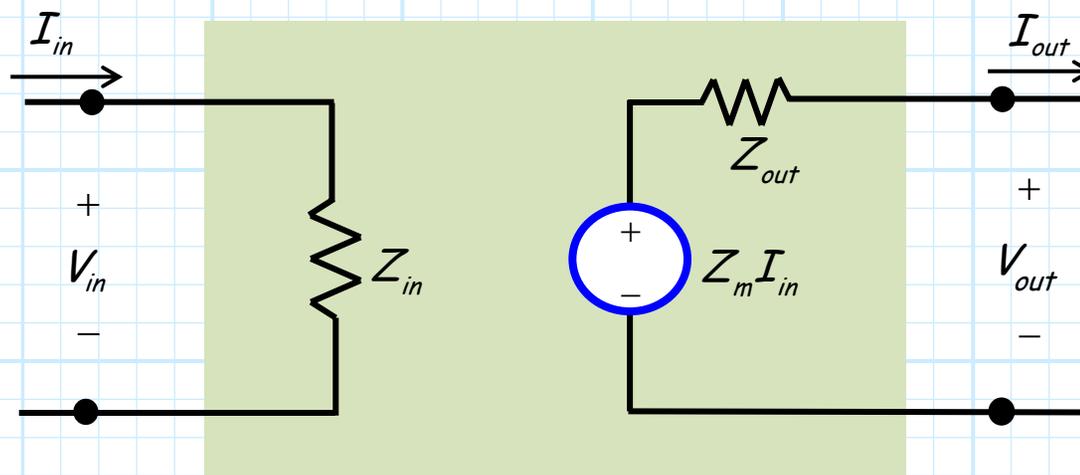
$$V_g = Z_m I_{in} \quad \text{and} \quad Z_g = Z_{out}$$



$$V_{out} = V_g - Z_g I_{out}$$

An equivalent circuit model

We can combine these two observations to form an **equivalent circuit model** of an (unilateral) amplifier:



$$V_{in} = Z_{in} I_{in}$$

$$V_{out} = Z_m I_{in} - Z_{out} I_{out}$$

Note in this model, the output of the amp is a **dependent** Thevenin's source—dependent on the **input current**!

Let's make the model more useful

Q: *So, do we **always** use **this** equivalent circuit to model an amplifier?*

A: Um, actually **no**.

The **truth** is that we EE's rarely use this equivalent circuit (not that there's anything wrong with it!).

Instead, the equivalent circuit we use involves a **slight modification** of the model above.

Relate the input voltage to output voltage

To see this modification, we insert the **first** (i.e., input) equation, expressed as:

$$I_{in} = \frac{V_{in}}{Z_{in}}$$

into the **second** (i.e., output) equation:

$$\begin{aligned} V_{out} &= Z_m I_{in} - Z_{out} I_{out} \\ &= \left(\frac{Z_m}{Z_{in}} \right) V_{in} - Z_{out} I_{out} \end{aligned}$$

Thus, the **open-circuit output voltage** can alternatively be expressed in terms of the **input voltage**!

$$V_{out}^{oc} = \left(\frac{Z_m}{Z_{in}} \right) V_{in}$$

Note the ratio Z_m/Z_{in} is unitless (a coefficient!).

Open-circuit voltage gain

This coefficient is known as the **open-circuit voltage gain** A_{vo} of an amplifier:

$$A_{vo}(\omega) \doteq \frac{V_{out}^{oc}}{V_{in}} = \frac{Z_m}{Z_{in}} = \frac{Z_{21}}{Z_{11}}$$

The **open-circuit voltage gain** $A_{vo}(\omega)$ is perhaps the most **important** of all amplifier parameters.

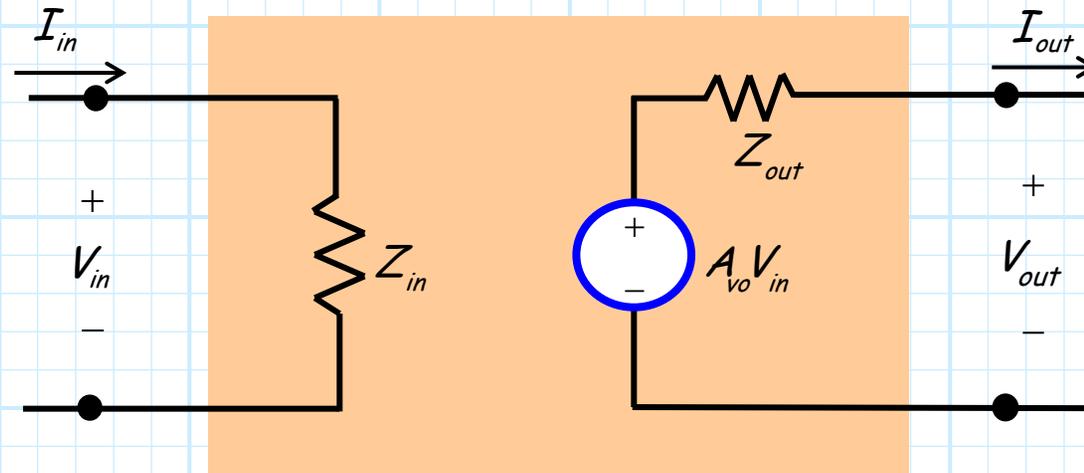
To see why, consider the amplifier equations in terms of this voltage gain:

$$V_{in} = Z_{in} I_{in}$$

$$V_{out} = A_{vo} V_{in} - Z_{out} I_{out}$$

A more "useful" equivalent circuit

The equivalent circuit described by these equations is:



In this circuit model, the output Thevenin's source is again dependent—but now it's dependent on the input **voltage**!

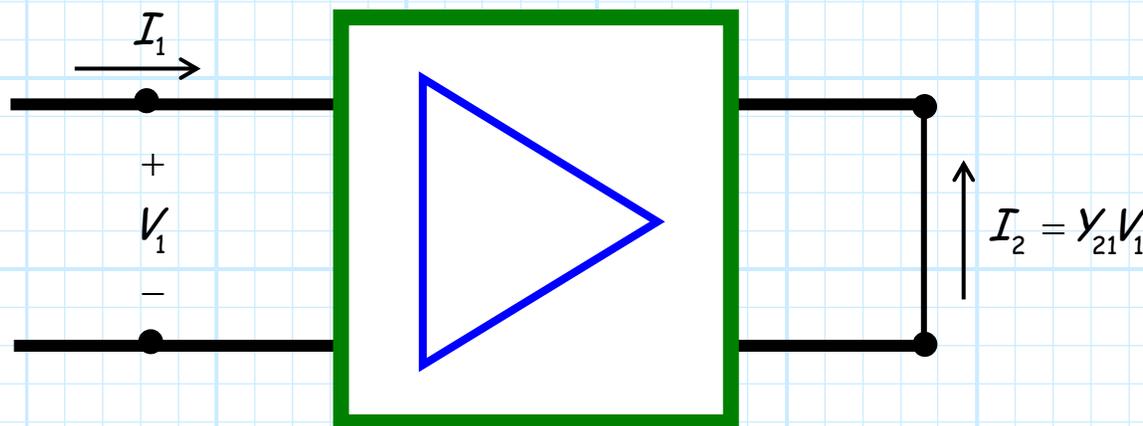
Thus, in this model, the input **voltage** and output **voltage** are more **directly related**.

Now let's consider admittance parameters

Q: *Are these the **only** two ways to model a unilateral amplifier?*

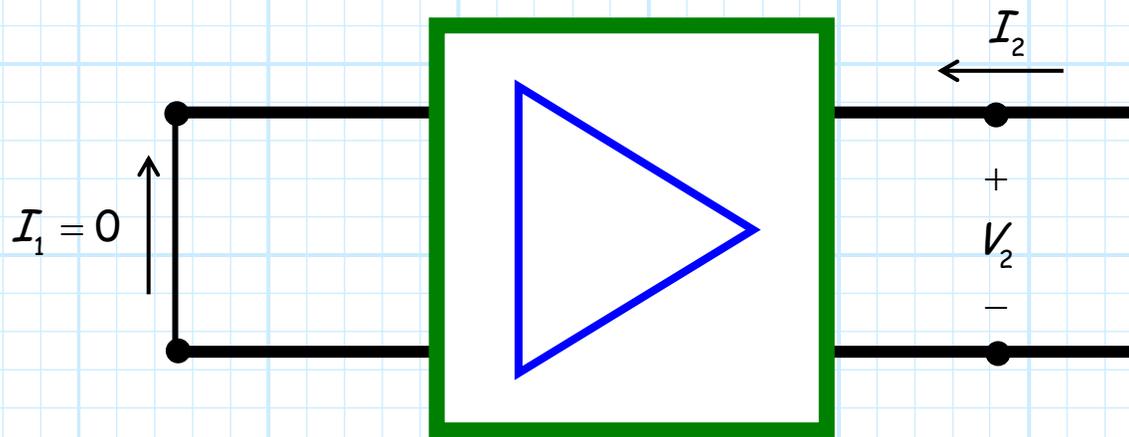
A: Hardly! Consider now **admittance parameters**.

A **voltage** on the **input** port of an amplifier will indeed produce a **short-circuit output current**:



The unilateral amplifier

However, since amplifiers are **not** reciprocal, placing the **same** voltage at the output will **not** create the equal current at the input—in fact, it will produce **no current at all!**



This again shows that amplifiers are **unilateral** devices, and so we find that the trans-admittance parameter Y_{12} is **zero**:

$$Y_{12}(\omega) = 0 \quad (\text{for amplifiers})$$

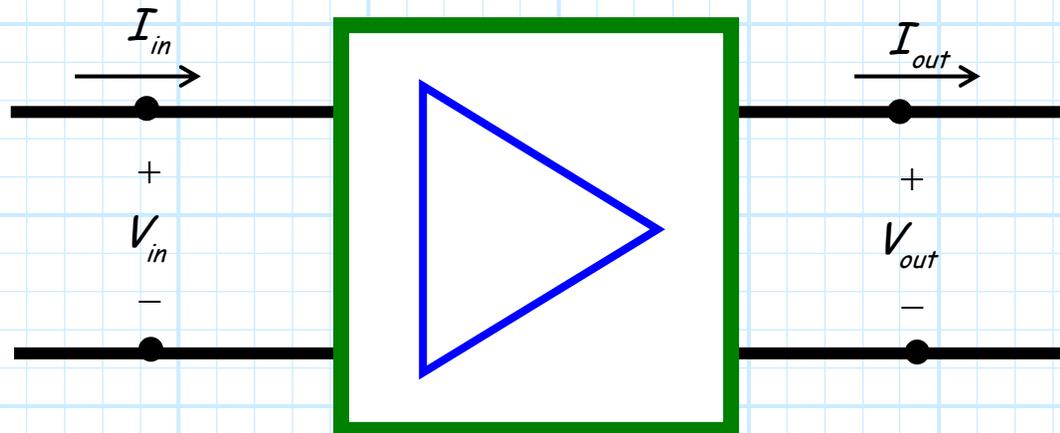
In terms of our new notation

Thus, the two equations using admittance parameters simplify to:

$$I_1 = Y_{11} V_1$$

$$I_2 = Y_{21} V_1 + Y_{22} V_2$$

with the same definitions of input and output current/voltage used previously:



$$I_{in} = Y_{11} V_{in}$$

$$I_{out} = -Y_{21} V_{in} - Y_{22} V_{out}$$

Input admittance

As with impedance parameters, it is apparent from this result that the **input** port is **independent** from the **output**.

Specifically, an **input admittance** can be defined as:

$$y_{in}(\omega) \doteq \frac{I_{in}(\omega)}{V_{in}(\omega)} = y_{11}(\omega)$$

Note that the **input admittance** of an amplifier is simply the **inverse** of the **input impedance**:

$$y_{in}(\omega) = \frac{I_{in}(\omega)}{V_{in}(\omega)} = \frac{1}{Z_{in}(\omega)}$$

And from this we can conclude that for a **unilateral** amplifier (but **only** because it's unilateral!):

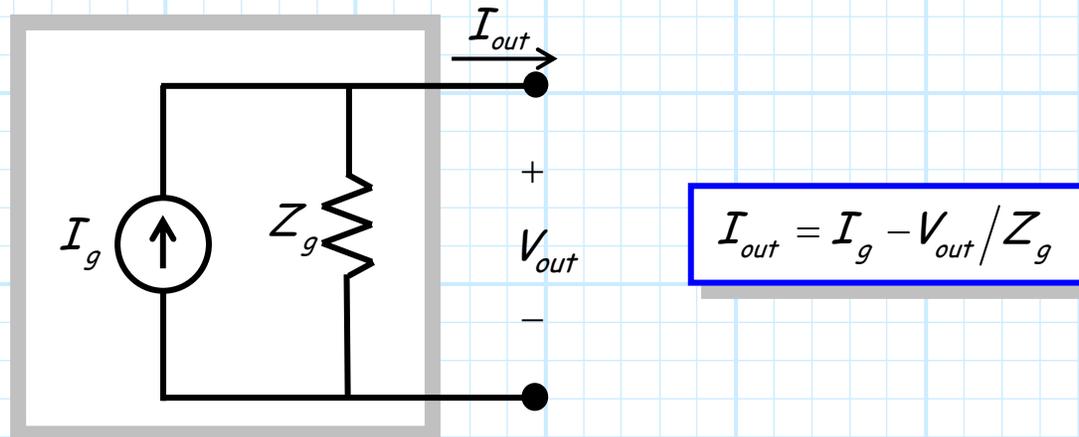
$$y_{11} = \frac{1}{Z_{11}}$$

Looks like a Norton's source!

Likewise, the **second** amplifier equation:

$$I_{out} = -y_{21} V_{in} - y_{22} V_{out}$$

is of the form of a **Norton's source**:



$$I_{out} = I_g - V_{out} / Z_g$$

where:

$$I_g = -y_{21} V_{in} \quad \text{and} \quad Z_g = \frac{1}{y_{22}}$$

Short-circuit trans-admittance

More specifically, we can define a **short-circuit trans-admittance**:

$$y_m \doteq -Y_{21}$$

and an **output impedance**:

$$Z_{out} \doteq \frac{1}{y_{22}}$$

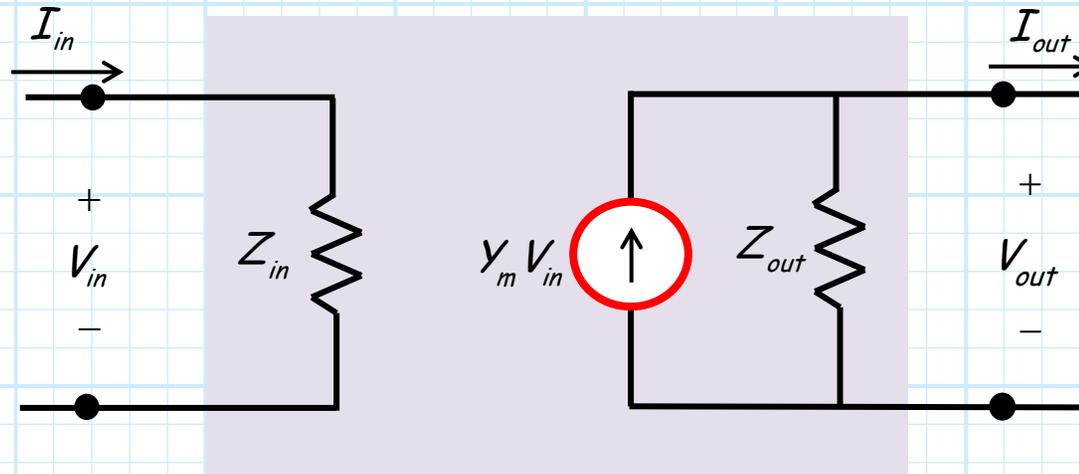
so that the amplifier equations are **now**:

$$I_{in} = V_{in} / Z_{in}$$

$$I_{out} = y_m V_{in} - V_{out} / Z_{out}$$

Yet another equivalent circuit model

The equivalent circuit described by **these** equations is:



Note in this model, the output of the amp is a **dependent** Norton's source—dependent on the **input voltage**.

However, this particular amplifier model is likewise **seldom** used.



Short-circuit current gain

Instead, we again insert the **input** equation:

$$V_{in} = \frac{I_{in}}{Y_{11}}$$

Into the **output** equation:

$$\begin{aligned} I_{out} &= -Y_{21} V_{in} - V_{out} / Z_{out} \\ &= -\left(\frac{Y_{21}}{Y_{11}} \right) I_{in} - V_{out} / Z_{out} \end{aligned}$$

Note the ratio $-Y_{21}/Y_{11}$ is unitless (a coefficient!).

This coefficient is known as the **short-circuit current gain** A_{is} of an amplifier:

$$A_{is}(\omega) \doteq \frac{I_{out}^{sc}}{I_{in}} = \frac{Y_m}{Y_{11}} = -\frac{Y_{21}}{Y_{11}}$$

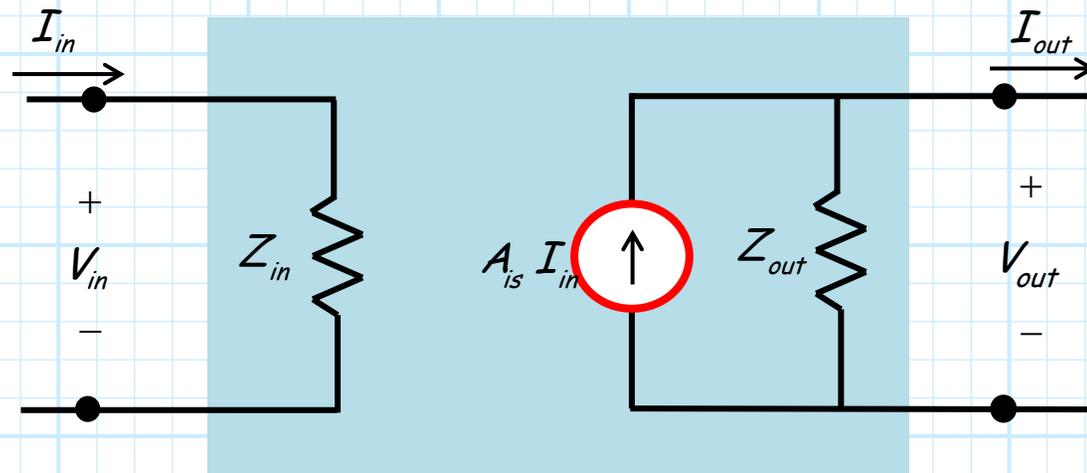
A useful equivalent circuit model

Thus, we can also express the amplifier port equations as:

$$I_{in} = V_{in} / Z_{in}$$

$$I_{out} = A_{is} I_{in} - V_{out} / Z_{out}$$

So, the equivalent circuit described by **these** equations is the **last of four** we shall consider:

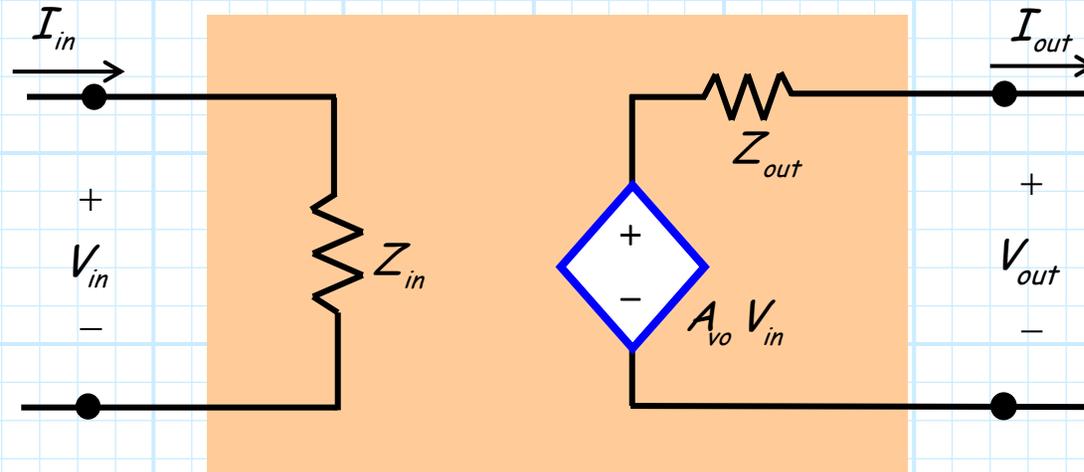


In this circuit model, the output Norton's source is again dependent—but now it's dependent on the input **current**!

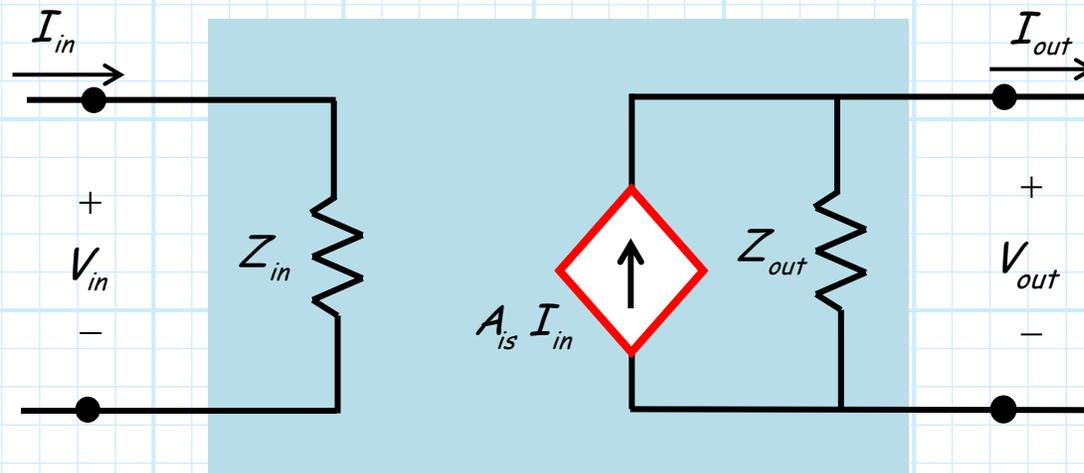
Thus, in this model, the input **current** and output **current** are more directly related.

Circuit Models for Amplifiers

The two most important amplifier circuit models explicitly use the **open-circuit voltage gain** A_{vo} :



And the **short-circuit current gain** A_{is} :



Just three values describe all!

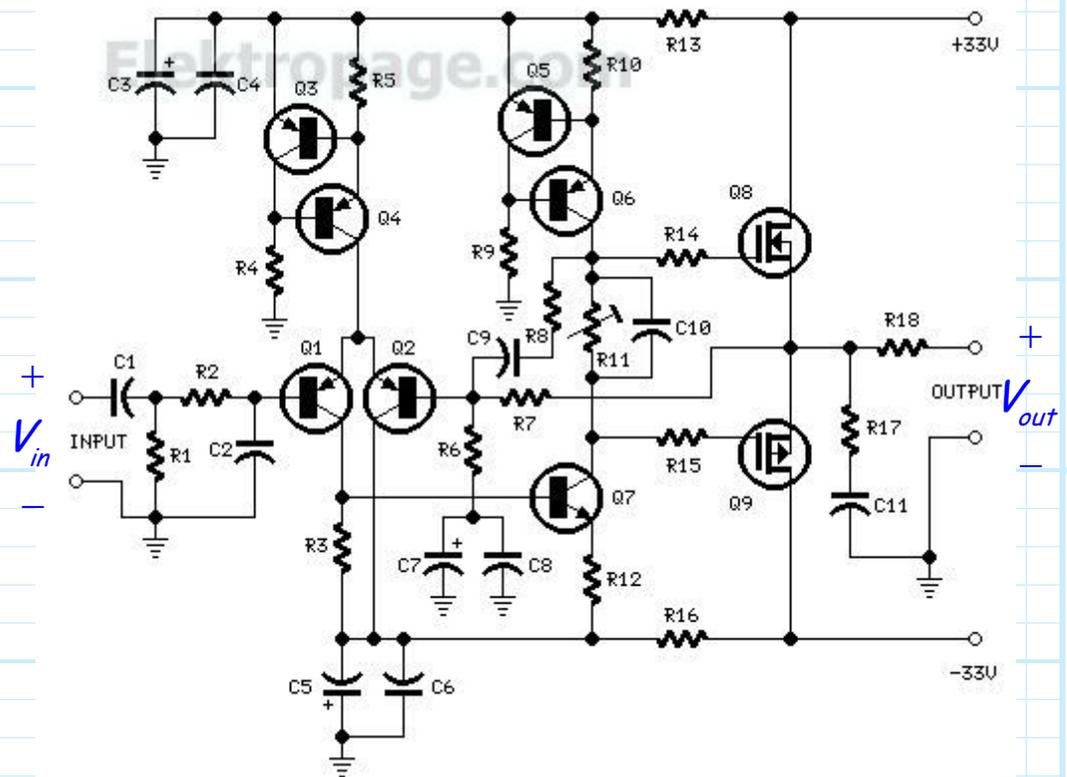
In addition, each equivalent circuit model uses the **same** two impedance values—the **input** impedance Z_{in} and **output** impedance Z_{out} .

Q: *So what are these models good for?*

A: Say we wish to analyze a circuit in which an amplifier is but **one** component.

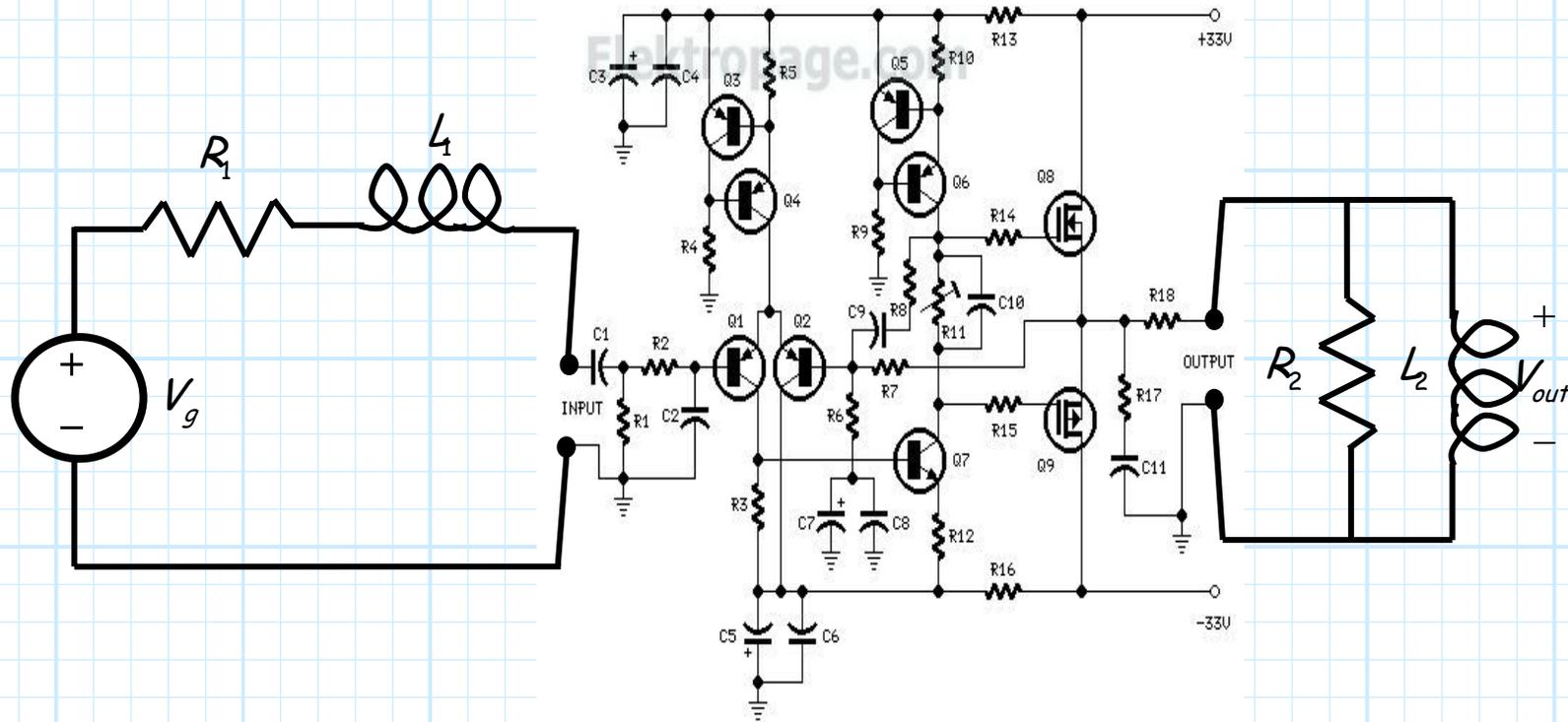
Instead of needing to analyze the **entire** amplifier circuit, we can analyze the circuit using the (far) **simpler** equivalent circuit model.

For **example**, consider **this** audio amplifier design:



This might be on the final

Say we wish to connect a **source** (e.g., microphone) to its **input**, and a **load** (e.g., speaker) to its **output**:



Let's say on the **EECS 412 final**, I ask you to determine V_{out} in the circuit above.

I'm not quite the jerk I appear to be!

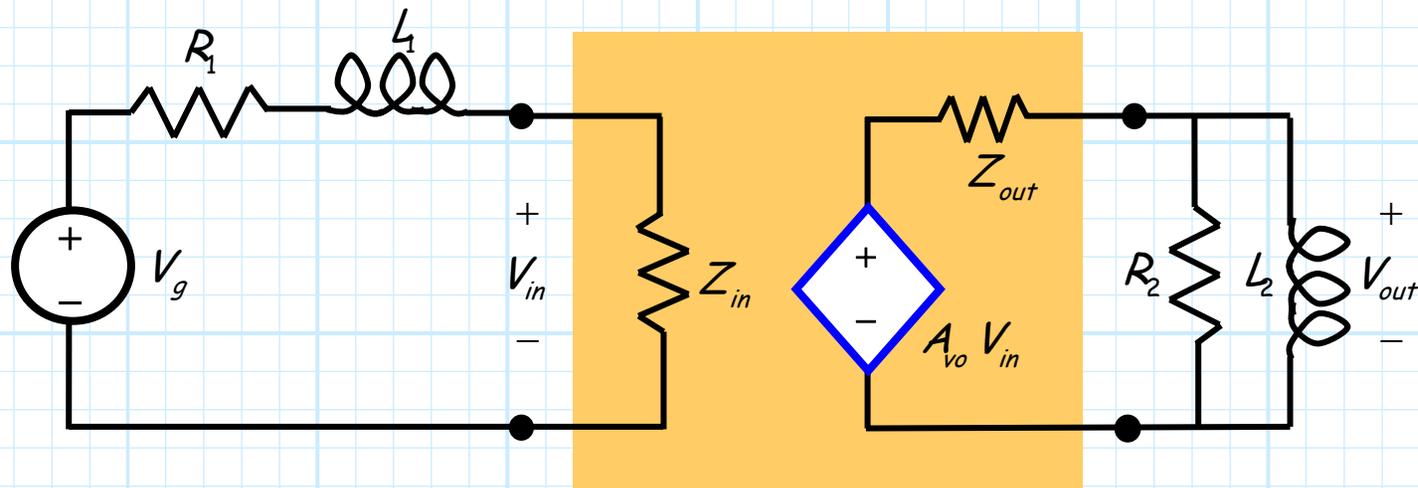


Q: Yikes! How could we *possibly* analyze this circuit on an exam—it would take way *too* much time (not to mention way *too* many pages of work)?

A: Perhaps, but let's say that I also provide you with the amplifier input impedance Z_{in} , output impedance Z_{out} , and open-circuit voltage gain A_{vo} .

You thus know **everything** there is to know about the amplifier!

Just replace the amplifier with its **equivalent circuit**:



The relationship between input and output voltages

From **input** circuit, we can conclude (with a little help from voltage division):

$$V_{in} = V_g \left(\frac{Z_{in}}{R_1 + j\omega L_1 + Z_{in}} \right)$$

And the **output** circuit is likewise:

$$V_{out} = A_{vo} V_{in} \left(\frac{R_2 \parallel j\omega L_2}{Z_{out} + R_2 \parallel j\omega L_2} \right)$$

where:

$$R_2 \parallel j\omega L_2 = \frac{j\omega R_2 L_2}{R_2 + j\omega L_2}$$

The output is not open-circuited!

Q: Wait! I *thought* we could determine the output voltage from the input voltage by simply multiplying by the voltage gain A_{vo} . I am **certain** that you told us:

$$V_{out}^{oc} = A_{vo} V_{in}$$

A: I did tell you that! And this expression is **exactly correct**.

However, the voltage V_{out}^{oc} is the **open-circuit** output voltage of the amplifier—in **this** circuit (like most amplifier circuits!), the output is **not open!**

Hence $V_{out} \neq V_{out}^{oc}$, and so :

$$\begin{aligned} V_{out} &= A_{vo} V_{in} \left(\frac{R_2 \parallel j\omega L_2}{Z_{out} + R_2 \parallel j\omega L_2} \right) \\ &= V_{out}^{oc} \left(\frac{R_2 \parallel j\omega L_2}{Z_{out} + R_2 \parallel j\omega L_2} \right) \\ &\neq V_{out}^{oc} \end{aligned}$$

We can define a voltage gain

Now, combining the two expressions, we have our **answer**:

$$\begin{aligned} V_{out} &= V_g A_{vo} \left(\frac{Z_{in}}{R_1 + j\omega L_1 + Z_{in}} \right) \left(\frac{R_2 \parallel j\omega L_2}{Z_{out} + R_2 \parallel j\omega L_2} \right) \\ &= A_{vo} V_g \left(\frac{Z_{in}}{R_1 + j\omega L_1 + Z_{in}} \right) \left(\frac{j\omega R_2 L_2}{Z_{out} (R_2 + j\omega L_2) + j\omega R_2 L_2} \right) \end{aligned}$$

Now, be aware that we can (and often do!) **define** a voltage gain A_v , a value that is different from the **open-circuit** voltage gain of the **amplifier**.

For instance, in the above circuit example we could **define** a voltage gain as the ratio of the input voltage V_{in} and the output voltage V_{out} :

$$A_v \doteq \frac{V_{out}}{V_{in}} = A_{vo} \left(\frac{R_2 \parallel j\omega L_2}{Z_{out} + R_2 \parallel j\omega L_2} \right) = A_{vo} \left(\frac{j\omega R_2 L_2}{Z_{out} (R_2 + j\omega L_2) + j\omega R_2 L_2} \right)$$

Or we can define a different gain

Or, we could alternatively **define** voltage gain as the ratio of the source voltage V_g and the output voltage V_{out} :

$$A_v \doteq \frac{V_{out}}{V_g} = A_{vo} \left(\frac{Z_{in}}{R_1 + j\omega L_1 + Z_{in}} \right) \left(\frac{j\omega R_2 L_2}{Z_{out} (R_2 + j\omega L_2) + j\omega R_2 L_2} \right)$$

Q: *Yikes! Which result is correct; which voltage gain is "the" voltage gain?*

A: Both are!

We can **define** a voltage gain A_v in **any** manner that is **useful** to us. However, we must make this definition explicit—**precisely** what two voltages are involved in the definition?

→ **No** voltage gain A_v is "the" voltage gain!

Note that the open-circuit voltage gain A_{vo} is a parameter of the **amplifier**—and of the amplifier **only**!

The open-circuit gain is the amplifier gain

Contrast A_{vo} to the two voltage gains defined above (i.e., V_{out}/V_{in} and V_{out}/V_g).

In each case, the result—of course—depends on **amplifier** parameters (A_{vo}, Z_{in}, Z_{out}).

However, the results **likewise** depend on the devices (source and load) **attached** to the amplifier (e.g., L_1, R_1, L_2, R_2).

→ The only **amplifier** voltage gain is its **open-circuit** voltage gain A_{vo} !

The low-frequency model

Now, let's switch gears and consider **low-frequency** (e.g., audio and video) applications.

At these frequencies, parasitic elements are typically **too small** to have any practical significance.

Additionally, low-frequency circuits **frequently** employ **no** reactive circuit elements (no capacitor or inductors).

As a result, we find that the input and output **impedances** exhibit almost **no imaginary** (i.e., reactive) components:

$$Z_{in}(\omega) \cong R_{in} + j0$$

$$Z_{out}(\omega) \cong R_{out} + j0$$

We can express this in the time domain

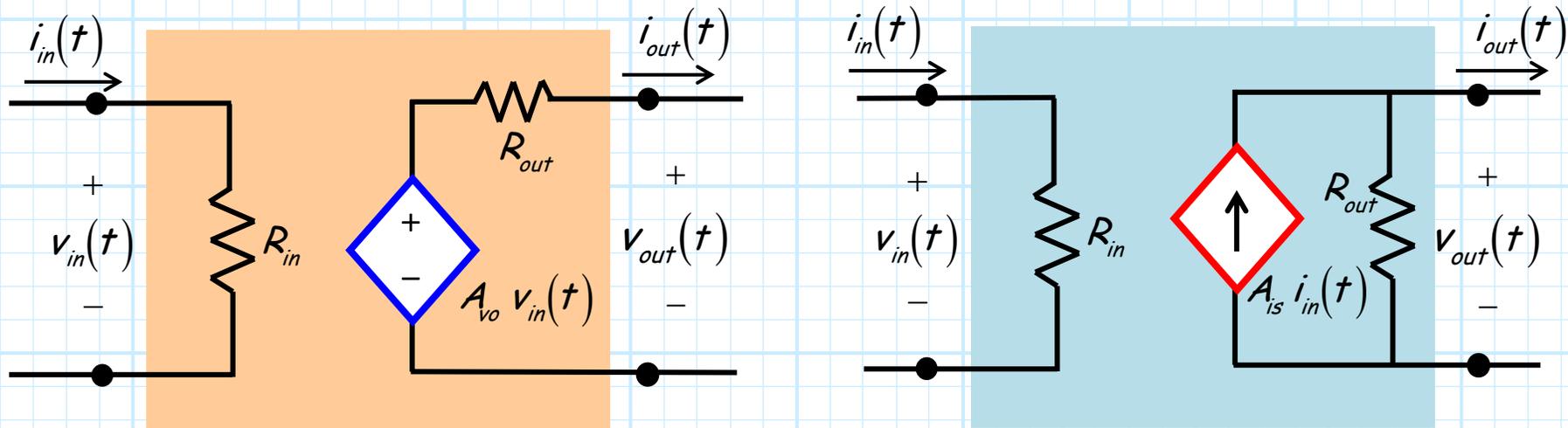
Likewise, the voltage and current **gains** of the amplifier are (almost) purely **real**:

$$A_{vo}(\omega) \cong A_{vo} + j0$$

$$A_{is}(\omega) \cong A_{is} + j0$$

Note that these real values can be **positive** or **negative**.

The amplifier **circuit models** can thus be **simplified**—to the point that we can easily consider arbitrary **time-domain** signals (e.g., $v_{in}(t)$ or $i_{out}(t)$):



All real-valued

For this case, we find that the (approximate) relationships between the input and output are that of an **ideal** amplifier:

$$v_{out}^{oc}(t) = \int_{-\infty}^t A_{vo} \delta(t-t') v_{in}(t') dt' = A_{vo} v_{in}(t)$$

$$i_{out}^{sc}(t) = \int_{-\infty}^t A_{is} \delta(t-t') i_{in}(t') dt' = A_{is} i_{in}(t)$$

Specifically, we find that for these low-frequency **models**:

$$R_{in} = \frac{v_{in}(t)}{i_{in}(t)}$$

$$R_{out} = \frac{v_{out}^{oc}(t)}{i_{out}^{sc}(t)}$$

$$A_{vo} = \frac{v_{out}^{oc}(t)}{v_{in}(t)}$$

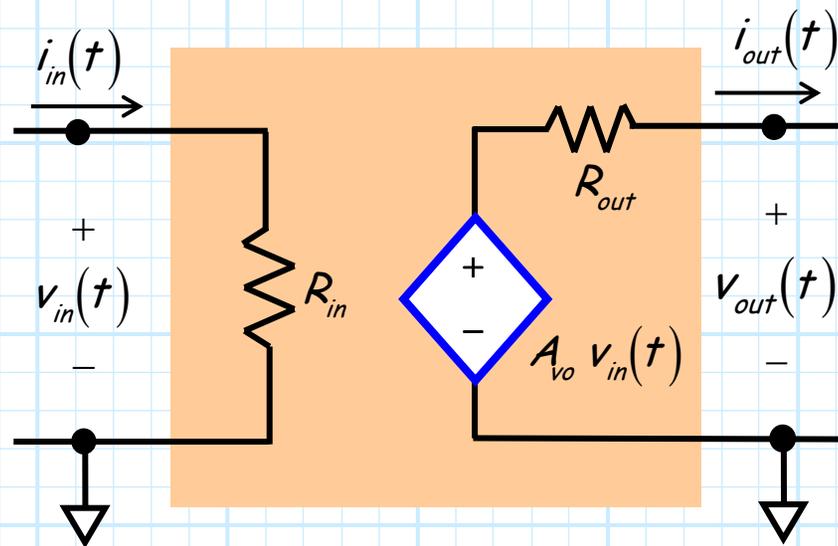
$$A_{is} = \frac{i_{out}^{sc}(t)}{i_{in}(t)}$$

One important **caveat** here; this "low-frequency" model is applicable only for **input signals** that are **likewise** low-frequency—the input signal spectrum must **not** extend beyond the amplifier **bandwidth**.

Voltage is referenced to ground potential

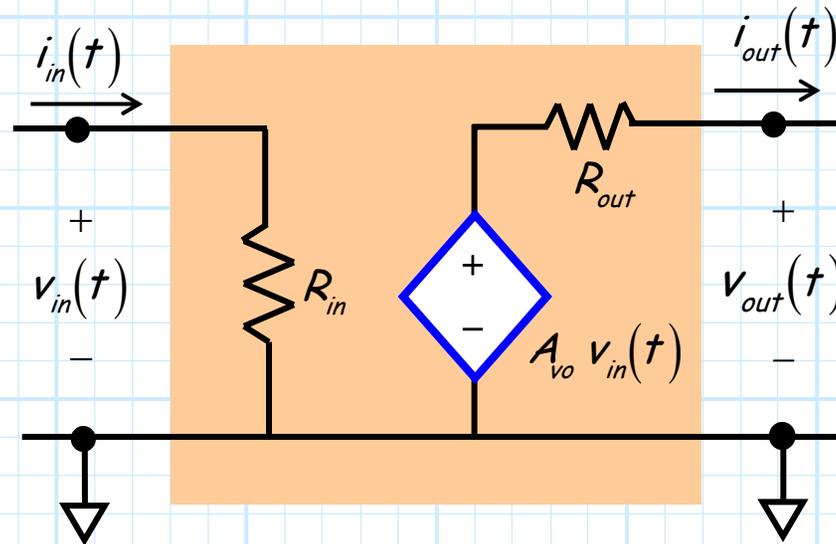
Now one last topic.

Frequently, both the input and output **voltages** are expressed with respect to **ground potential**, a situation expressed in the circuit **model** as:

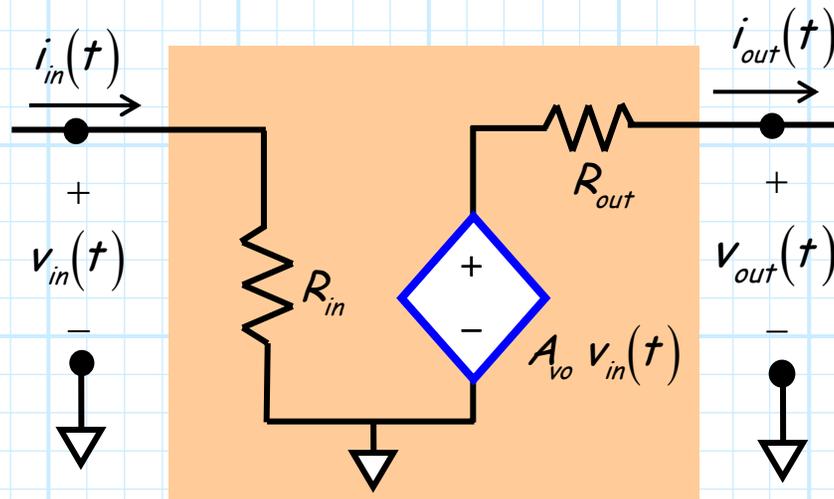


You'll often see this notation

Now, two nodes at ground potential are two nodes that are **connected** together!
Thus, an **equivalent** model to the one above is:



Which is **generally** simplified to this model:



Current and Voltage Amplifiers

Q: *I'll admit to being **dog-gone** confused.*

*You say that **every** amplifier can be described **equally** well in terms of **either** its open-circuit voltage gain A_{vo} , or its short-circuit current gain A_{is} .*

*Yet, amps I have seen are denoted **specifically** as either a dad-gum **current** amplifier or a gul-darn **voltage** amplifier.*

***Are** voltage and current amplifiers **separate** devices, and if so, **what** are the differences between them?*



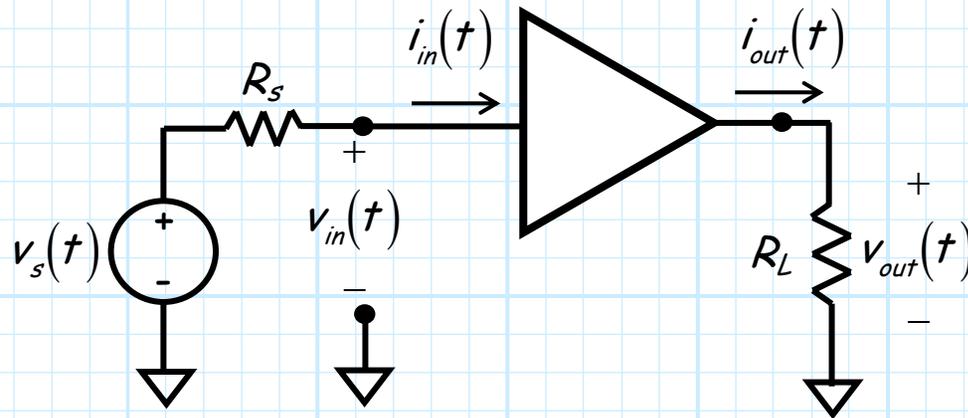
A: Any amplifier can be used as **either** a current amp or as a voltage amp.

However, we will find that an amp that works well as **one** does not generally work well as the **other**! Hence, we can in general **classify** amps as either voltage amps or current amps.

Define a gain

To see the difference we first need to provide some **definitions**.

First, consider the following circuit:



Q: Isn't that just A_{vo} ??

We **define** a voltage gain A_v as:

$$A_v \doteq \frac{v_{out}(t)}{v_s(t)}$$



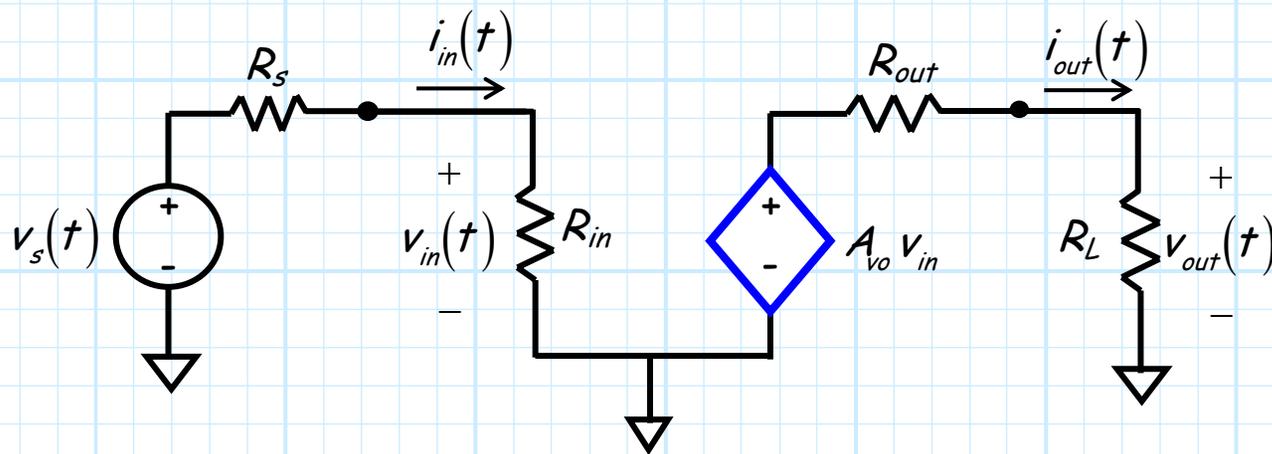
A: NO! Notice that the output of the amplifier is **not open circuited**.

This is what the model is for

Likewise, the **source** voltage v_s is **not** generally equal to the **input** voltage v_{in} .

We must use a **circuit model** to determine voltage gain A_v .

Although we can use **either** model, we will find it easier to analyze the **voltage** gain if we use the model with the dependent **voltage** source:



The result

Analyzing the **input** section of this circuit, we find:

$$v_{in} = \left(\frac{R_{in}}{R_s + R_{in}} \right) v_s$$

and analyzing the **output**:

$$v_{out} = \left(\frac{R_L}{R_{out} + R_L} \right) A_{vo} v_{in}$$

combining the two expressions we get:

$$v_{out} = \left(\frac{R_L}{R_{out} + R_L} \right) A_{vo} \left(\frac{R_{in}}{R_s + R_{in}} \right) v_s$$

and therefore the **voltage gain** A_v is:

$$A_v \doteq \frac{v_{out}(t)}{v_s(t)} = \left(\frac{R_L}{R_{out} + R_L} \right) A_{vo} \left(\frac{R_{in}}{R_s + R_{in}} \right)$$



How to maximize voltage gain

Note in the above expression that the first and third product terms are **limited**:

$$0 \leq \left(\frac{R_L}{R_{out} + R_L} \right) \leq 1 \quad \text{and} \quad 0 \leq \left(\frac{R_{in}}{R_s + R_{in}} \right) \leq 1$$

We find that each of these terms will approach their **maximum** value (i.e., one) when:

$$R_{out} \ll R_L \quad \text{and} \quad R_{in} \gg R_s$$

Thus, if the **input** resistance is very **large** ($\gg R_s$) and the **output** resistance is very **small** ($\ll R_L$), the voltage gain for this circuit will be **maximized** and have a value approximately **equal** to the **open-circuit voltage gain**!

$$v_o \approx A_{vo} v_s \quad \text{iff} \quad R_{out} \ll R_L \quad \text{and} \quad R_{in} \gg R_s$$

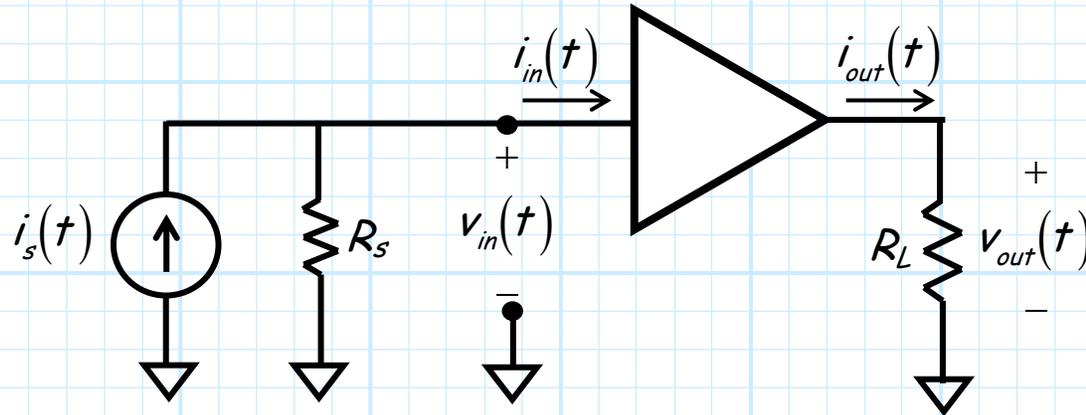
A good voltage amplifier

Thus, we can infer **three** characteristics of a good **voltage amplifier**:

1. Very large input resistance ($R_{in} \gg R_s$).
2. Very small output resistance ($R_{out} \ll R_L$).
3. Large open-circuit voltage gain ($A_{vo} \gg 1$).

Now for current gain

Now let's consider a **second** circuit:



We define **current gain** A_i as:

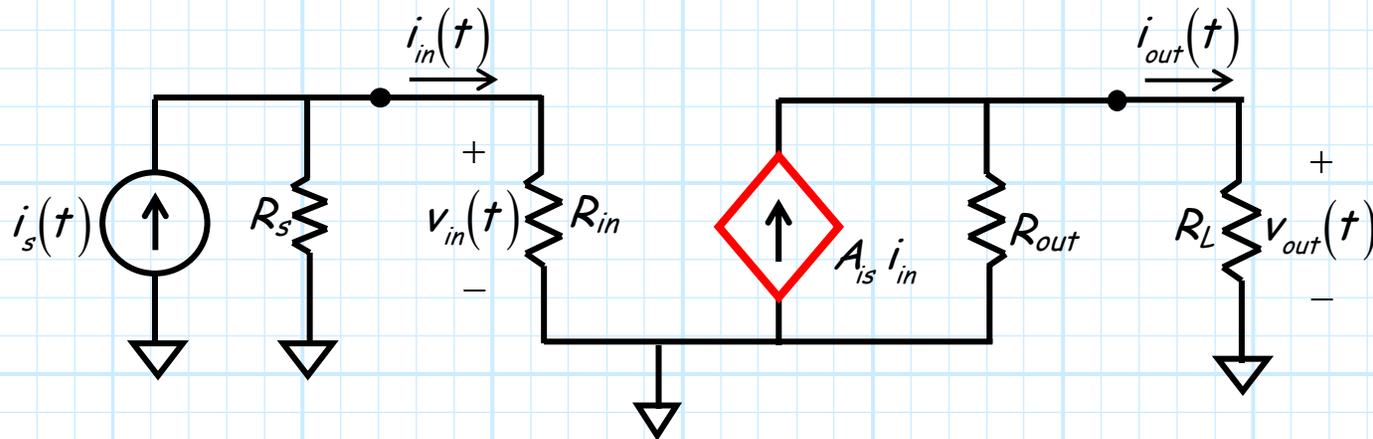
$$A_i \doteq \frac{i_{out}(t)}{i_s(t)}$$

Note that this gain is **not** equal to the **short-circuit** current gain A_{is} . This current gain A_i depends on the **source** and **load** resistances, as well as the amplifier parameters.

Therefore, we must use a **circuit model** to determine current gain A_i .

Use the other model

Although we can use **either** model, we will find it easier to analyze the **current** gain if we use the model with the dependent **current** source:



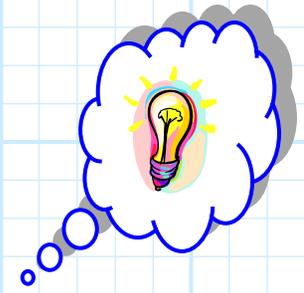
Analyzing the **input** section, we can use **current division** to determine:

$$i_{in} = \left(\frac{R_s}{R_s + R_{in}} \right) i_s$$

We likewise can use current division to analyze the **output** section:

$$i_{out} = \left(\frac{R_{out}}{R_{out} + R_L} \right) A_{is} i_{in}$$

How to maximize current gain



Combining these results, we find that:

$$i_{out} = \left(\frac{R_{out}}{R_{out} + R_L} \right) A_{is} \left(\frac{R_s}{R_s + R_{in}} \right) i_s$$

and therefore the **current gain** A_i is:

$$A_i \doteq \frac{i_o(t)}{i_s(t)} = \left(\frac{R_{out}}{R_{out} + R_L} \right) A_{is} \left(\frac{R_s}{R_s + R_{in}} \right)$$



Note in the above expression that the first and third product terms are **limited**:

$$0 \leq \left(\frac{R_{out}}{R_{out} + R_L} \right) \leq 1 \quad \text{and} \quad 0 \leq \left(\frac{R_s}{R_s + R_{in}} \right) \leq 1$$

We find that each of these terms will approach their **maximum** value (i.e., one) when:

$$R_{out} \gg R_L \quad \text{and} \quad R_{in} \ll R_s$$

The ideal current amp

Thus, if the **input** resistance is very **small** ($\ll R_s$) and the **output** resistance is very **large** ($\gg R_L$), the voltage gain for this circuit will be maximized and have a value approximately equal to the short-circuit current gain!

$$i_{out} \approx A_{is} i_s \quad \text{iff } R_{out} \gg R_L \text{ and } R_{in} \ll R_s$$

Thus, we can infer **three** characteristics of a good **current amplifier**:

1. Very **small input** resistance ($R_i \ll R_s$).
2. Very **large output** resistance ($R_o \gg R_L$).
3. Large short-circuit **current gain** ($A_{is} \gg 1$).

Note the ideal resistances are **opposite** to those of the ideal **voltage** amplifier!

You can trust ol' Roy!



*It's actually quite **simple**.*

*An amplifier with **low** input resistance and **high** output resistance will typically provide great **current** gain but lousy **voltage** gain.*

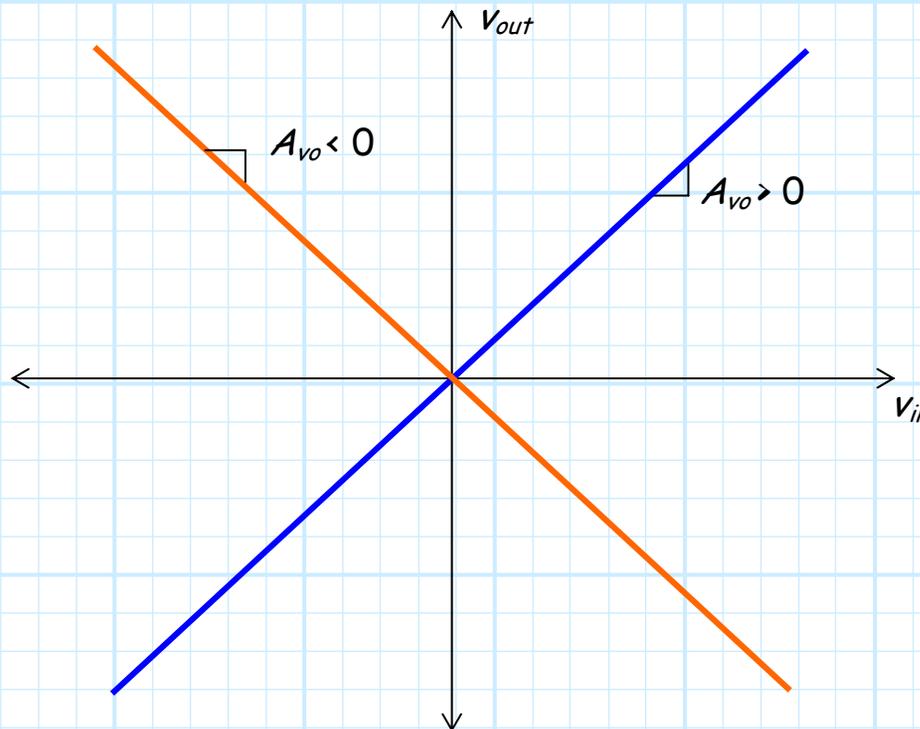
*Conversely, an amplifier with **high** input resistance and **low** output resistance will typically make a great **voltage** amplifier but a dog-gone poor **current** amp.*

Non-Linear Behavior of Amplifiers

Note that the **ideal** amplifier transfer function:

$$v_{out}^{oc}(t) = A_{vo} v_i(t)$$

is an equation of a **line** (with slope = A_{vo} and y -intercept = 0).



The output voltage is limited

This **ideal** transfer function implies that the **output voltage** can be **very large**, provided that the gain A_{vo} and the input voltage v_{in} are large.

However, we find in a "real" amplifier that there are **limits** on how large the output voltage can become.

The transfer function of an amplifier is more **accurately** expressed as:

$$v_{out}(t) = \begin{cases} L_+ & v_{in}(t) > L_+^{in} \\ A_{vo} v_{in}(t) & L_-^{in} < v_{in}(t) < L_+^{in} \\ L_- & v_{in}(t) < L_-^{in} \end{cases}$$

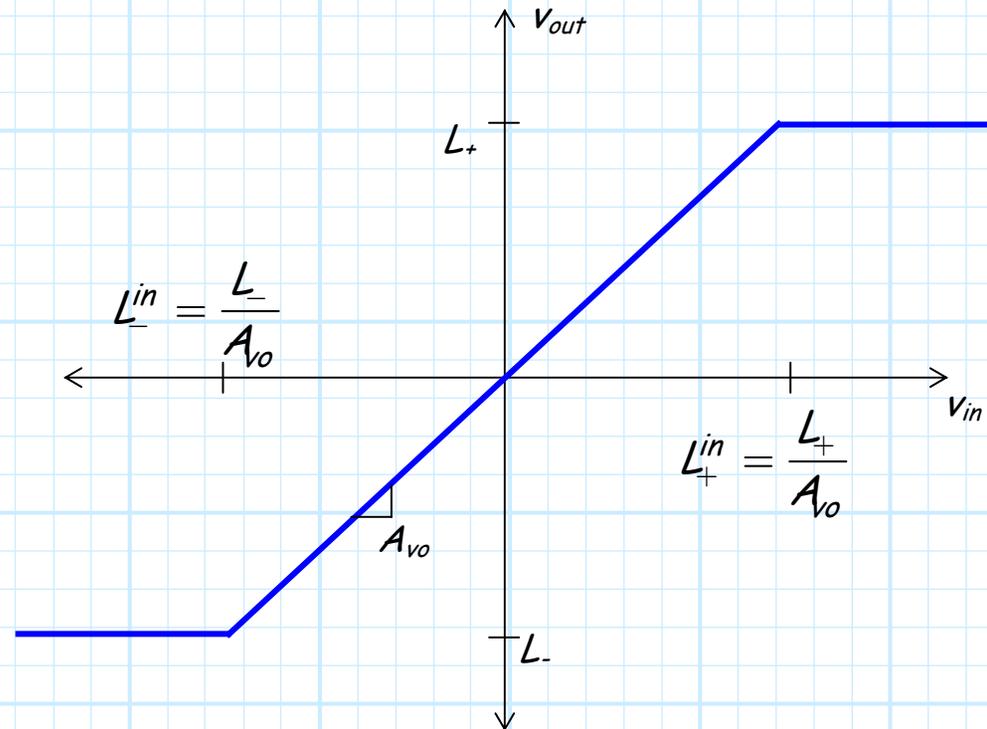
Amplifier saturation

This expression is shown **graphically** as:

This expression (and graph) shows that electronic amplifiers have a **maximum** and **minimum** output voltage (L_+ and L_-).

If the **input** voltage is either too large or too small (too negative), then the amplifier **output** voltage will be equal to either L_+ or L_- .

If $v_{out} = L_+$ or $v_{out} = L_-$, we say the amplifier is in **saturation** (or compression).



Make sure the input isn't too large!

Amplifier saturation occurs when the **input** voltage is **greater** than:

$$v_{in} > \frac{L_+}{A_{vo}} \doteq L_+^{in}$$

or when the **input** voltage is **less** than:

$$v_{in} < \frac{L_-}{A_{vo}} \doteq L_-^{in}$$

Often, we find that these voltage limits are **symmetric**, i.e.:

$$L_- = -L_+ \quad \text{and} \quad L_-^{in} = -L_+^{in}$$

For example, the output limits of an amplifier might be $L_+ = 15$ V and $L_- = -15$ V.

However, we find that these limits are also often **asymmetric** (e.g., $L_+ = +15$ V and $L_- = +5$ V).

Saturation: Who really cares?



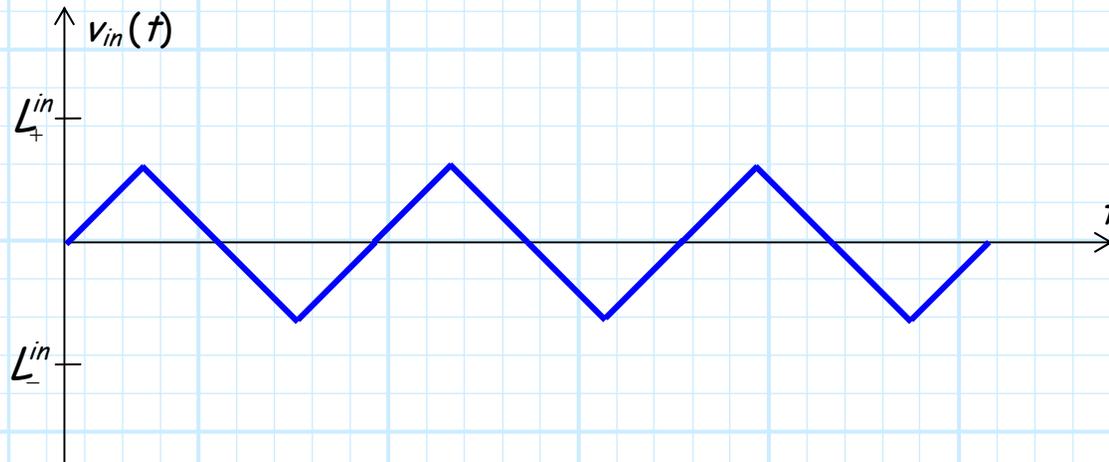
Q: *Why do we **care** if an amplifier saturates? Does it cause any **problems**, or otherwise result in performance **degradation**??*



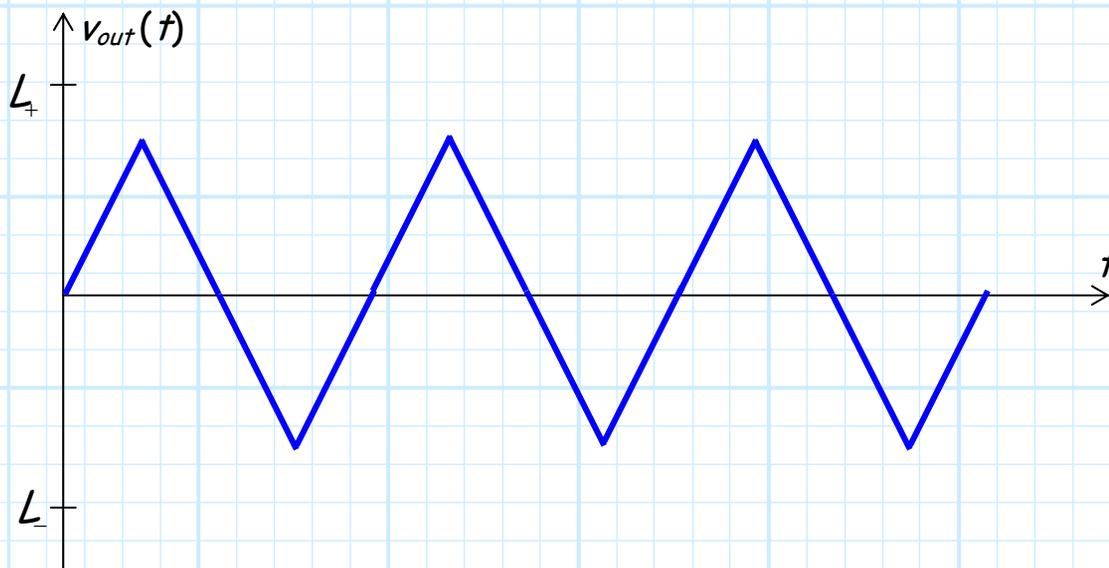
A: **Absolutely!** If an amplifier saturates—even momentarily—the unavoidable result will be a **distorted** output signal.

A distortion free example

For example, consider a case where the input to an amplifier is a **triangle wave**:

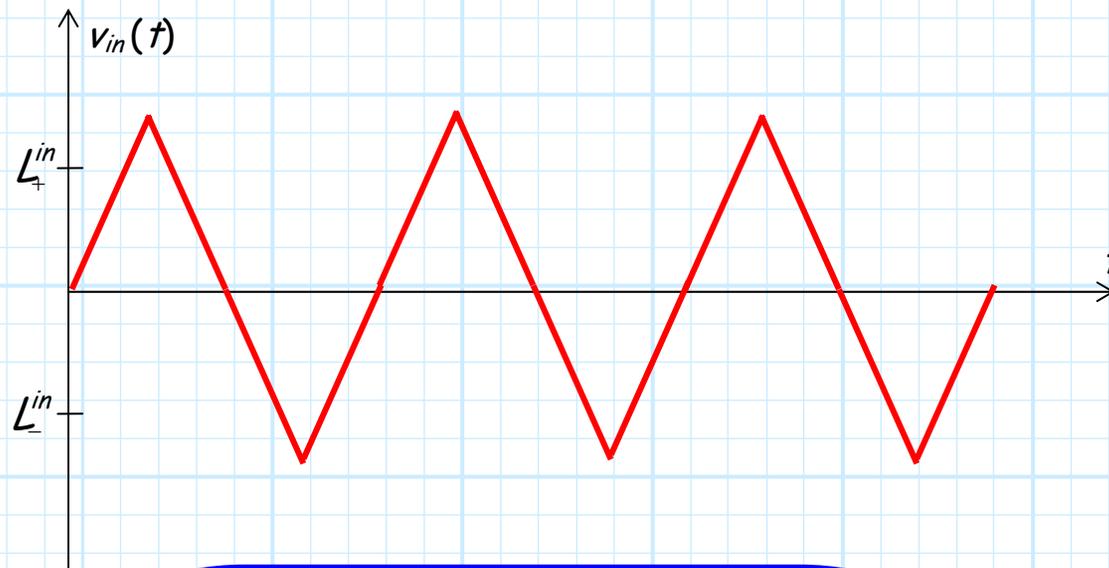


Since $L_-^{in} < v_{in}(t) < L_+^{in}$ for all time t , the **output** signal will be within the limits L_+ and L_- for all time t , and thus the amplifier output will be $v_{out}(t) = A_{vo} v_{in}(t)$:



The input is too darn big!

Consider now the case where the input signal is much **larger**, such that $v_{in}(t) > L_+^{in}$ and $v_{in}(t) < L_-^{in}$ for some time t (e.g., the input triangle wave **exceeds** the voltage limits L_+^{in} and L_-^{in} some of the time):

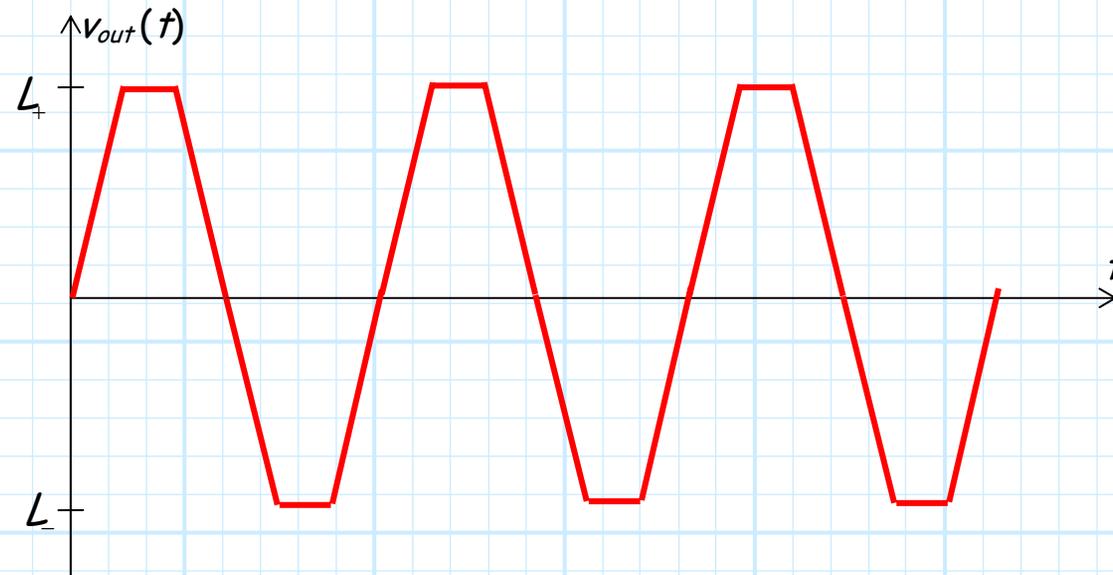


*This is precisely the situation about which I earlier expressed **caution**.*

*We now must experience the palpable agony of **signal distortion!***



Palpable agony



Note that this output signal is **not** a triangle wave!

For time t where $v_{in}(t) > L_+^{in}$ and $v_{in}(t) < L_-^{in}$, the value $A_{vo} v_{in}(t)$ is greater than L_+ and less than L_- , respectively.

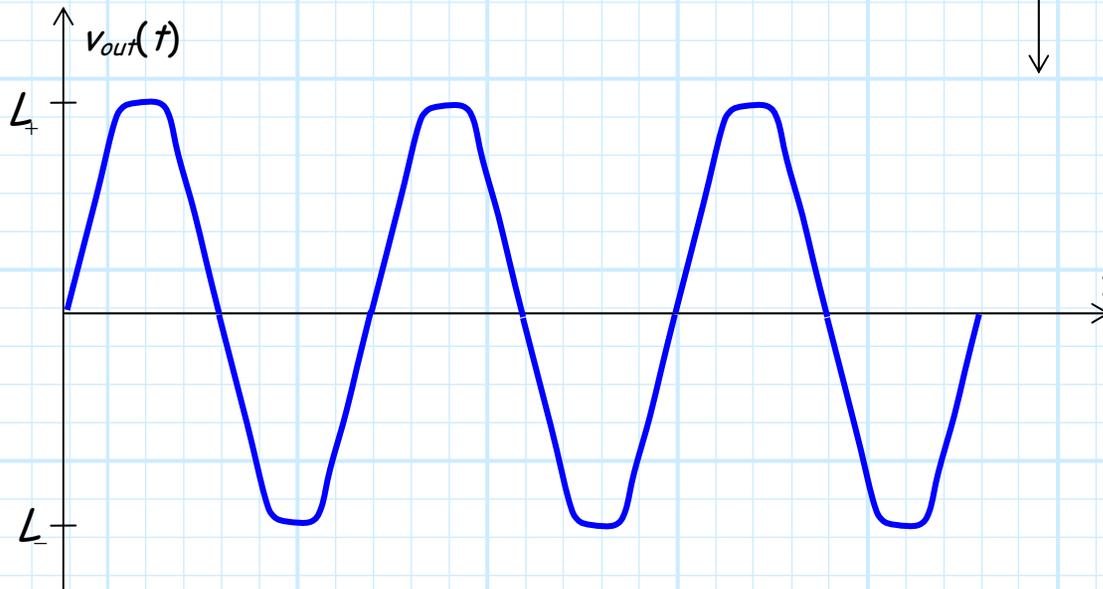
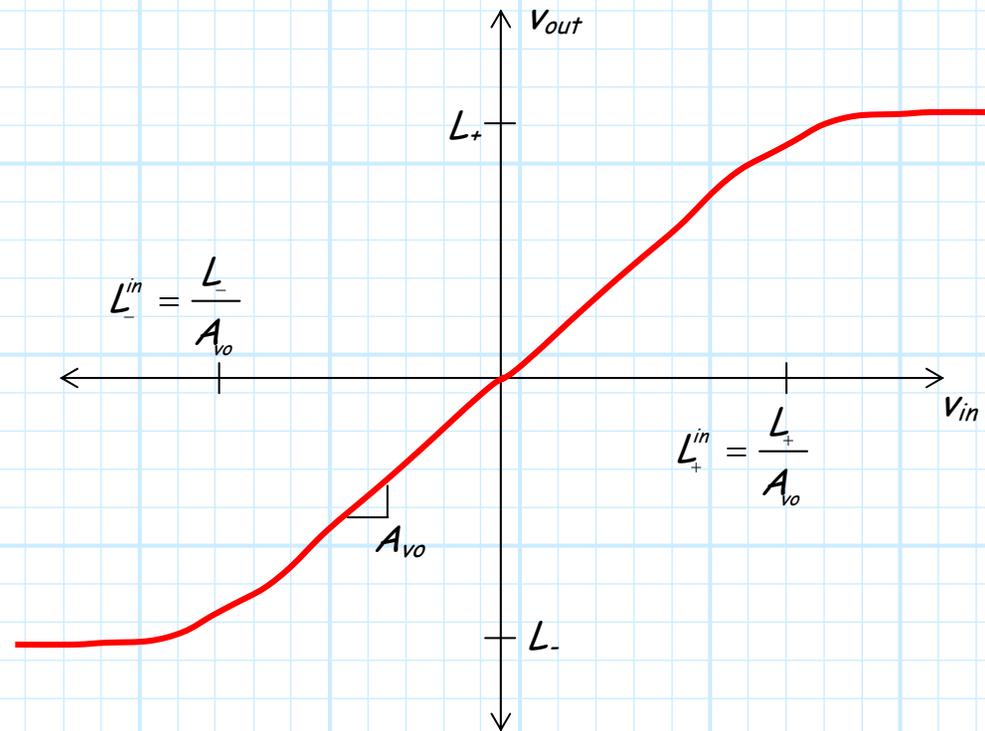
Thus, the output voltage is limited to $v_{out}(t) = L_+$ and $v_{out}(t) = L_-$ for these times.

As a result, we find that output $v_{out}(t)$ does **not** equal $A_{vo} v_{in}(t)$ —the output signal is **distorted!**

"Soft" Saturation

In reality, the **saturation** voltages L_+ , L_- , L_+^{in} , and L_-^{in} are not so **precisely** defined.

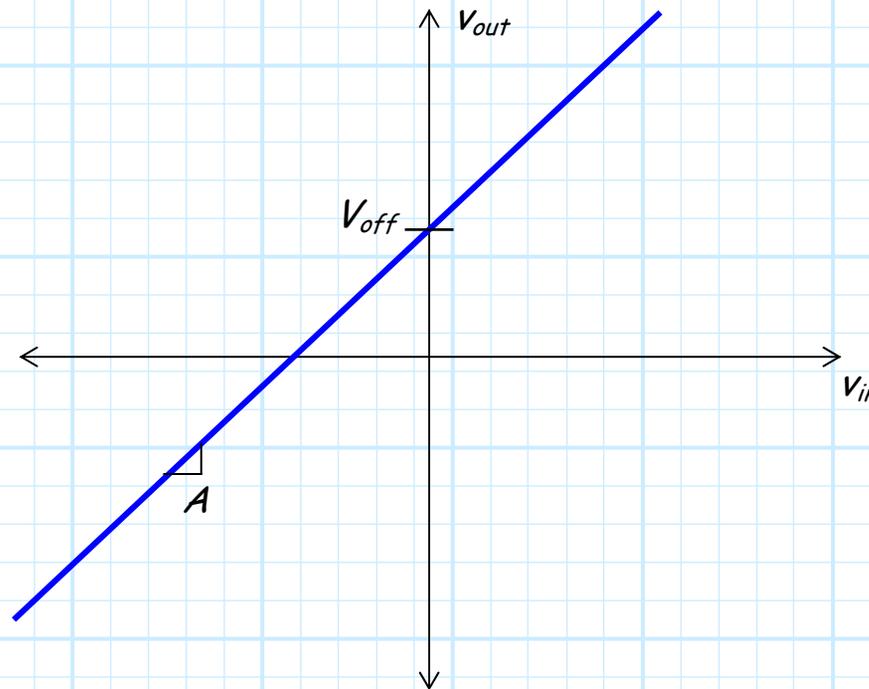
The transition from the linear amplifier region to the saturation region is **gradual**, and cannot be unambiguously defined at a precise point.



Yet another problem: DC offset

Now for another **non-linear** problem!

We will find that many amplifiers exhibit a **DC offset** (i.e., a **DC bias**) at their output.



How do we define gain?

The output of these amplifiers can be expressed as:

$$v_{out}(t) = A v_{in}(t) + V_{off}$$

where A and V_{off} are constants.

It is evident that if the input is **zero**, the output voltage will **not** be (zero, that is)!

i.e., $v_{out} = V_{off}$ if $v_{in} = 0$



Q: *Yikes! How do we determine the **gain** of such an amplifier?*

If: $v_{out}(t) = A v_{in}(t) + V_{off}$

then what is:

$$\frac{v_{out}(t)}{v_{in}(t)} = \text{?????}$$

*The **ratio** of the output voltage to input voltage is **not a constant!***

Calculus: is there anything it can't do?

A: The gain of any amplifier can be defined more precisely using the **derivative** operator:

$$A_{vo} \doteq \frac{dv_{out}}{dv_{in}}$$

Thus, for an amplifier with an output DC offset, we find the voltage gain to be:

$$A_{vo} = \frac{dv_{out}}{dv_{in}} = \frac{d(Av_{in} + V_{off})}{dv_{in}} = A$$

In other words, the gain of an amplifier is determined by the **slope** of the transfer function!

This sort of makes sense!

For an amplifier with **no** DC offset (i.e., $v_o = A_{vo} v_i$), it is easy to see that the gain is **likewise** determined from this definition:

$$A_{vo} = \frac{dv_{out}}{dv_{in}} = \frac{dA_{vo}v_{in}}{dv_{in}} = A_{vo}$$

*Hey, hey! This definition makes sense if you think about it—gain is the **change** of the output voltage with respect to a **change** at the input.*

*For example, of small change Δv_{in} at the **input** will result in a change of $A_{vo} \Delta v_{in}$ at the **output**.*

*If A_{vo} is **large**, this change at the output will be **large**!*



Both problems collide

OK, here's **another** problem.

The derivative of the transfer curve for **real** amplifiers will **not be a constant**.

We find that the gain of a amplifier will often be **dependent** on the input voltage!

The main reason for this is amplifier **saturation**.

Consider again the transfer function of an amplifier that **saturates**:

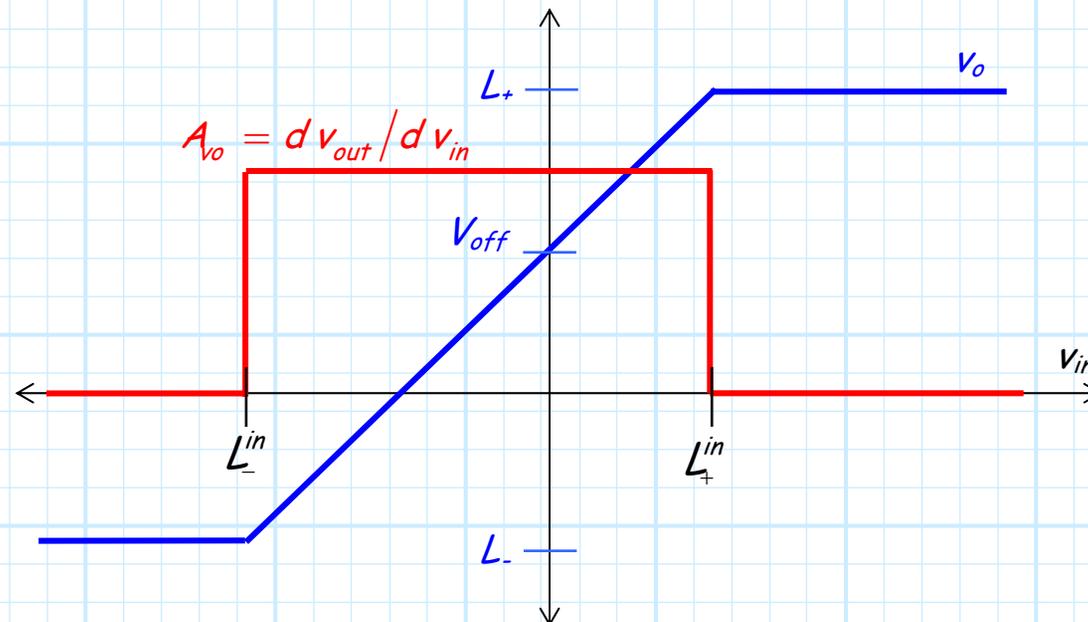
$$v_{out} = \begin{cases} L_+ & v_{in} > L_+^{in} \\ Av_{in} + V_{off} & L_-^{in} < v_i < L_+^{in} \\ L_- & v_{in} < L_-^{in} \end{cases}$$

Gain is a function of v_{in}

We find the **gain** of this amplifier by taking the **derivative** with respect to v_{in} :

$$A_{vo} = \frac{dv_{out}}{dv_{in}} = \begin{cases} 0 & v_{in} > L_+^{in} \\ A & L_-^{in} < v_{in} < L_+^{in} \\ 0 & v_{in} < L_-^{in} \end{cases}$$

Graphically, this result is:



You'll see this transfer function again!

Thus, the gain of this amplifier when in saturation is **zero**. A change in the input voltage will result in **no change** on the output—the output voltage will simply be $v_o = V_{\pm}$.

Again, the transition into saturation is **gradual** for real amplifiers.

In fact, we will find that many of the amplifiers studied in this class have a **transfer function** that looks something like this→

We will find that the voltage gain of many amplifiers is **dependent** on the input voltage.

Thus, a **DC bias** at the input of the amplifier is often required to **maximize** the amplifier gain.

