

Pure Data

Intermediate course



NINON DEVIS

https://ninon-io.github.io/

Made with love for ATIAM

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EXTERNALS Given by P. Esling

01 Processing

Signal normalization & various synthesis methods

Normalizing the Signal

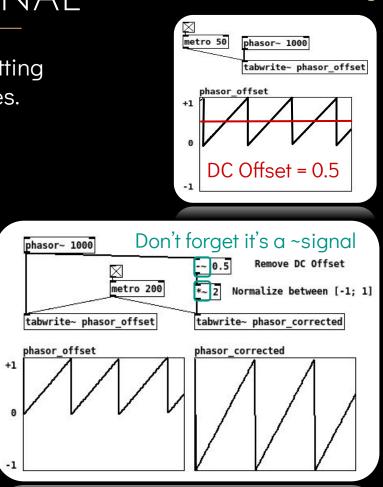
Goal: Obtaining the best dynamic range by fitting the gain of the signal into certain ranges.

Two Step-process:

- Removing the DC Offset
- Normalizing the signal

What is DC Offset ? Mean amplitude displacement from 0.

What is Normalization ? Adjust the gain to peak at the maximum the sound card allows before clipping: [-1, 1].



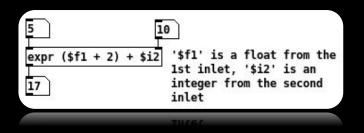
How to normalize this signal ?

NORMALIZING THE SIGNAL

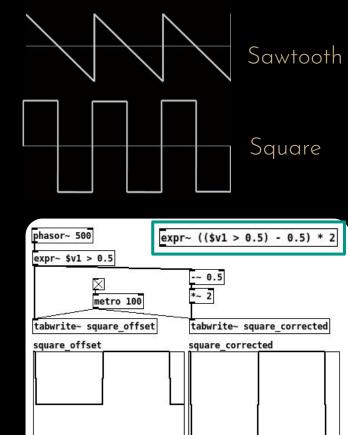
How to convert a sawtooth into a square?

- output = 1 if signal > 0.5 0 otherwise
- Introducing [expr] & [expr~]

The first inlet of [expr~] needs to be of type '\$v1' for a signal



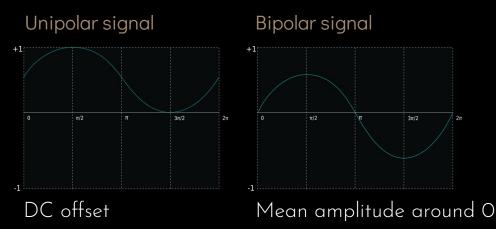
Select 'Polygon' in the properties of the array



MODULATION Multiplying audio signal

Difference between AM & RM

Modulation is typical in synthesis as it enriches the character of the sound and adds variance in timbre over time.



Reminder: Example for AM Carrier signal Modulating signal WWWWWWWWWWWWWWW Output R(t)ring modulation output A(t)amplitude modulation output C(t)carrier signal modulation signal M(t)

- Ring Modulation: multiplication of two bipolar signals by each others
- \rightarrow The frequency of the carrier signal is not present in the resulting sound.
- Amplitude Modulation: M is a unipolar modulator, typically between 0 and 1.
- \rightarrow The carrier frequency is preserved.

 $\overline{A(t)} = C(t) \times (M(t) + 1)$

 $R(t) = C(t) \times M(t)$

RING MODULATION

 $R(t) = C(t) \times M(t)$

We multiply 2 bipolar signals by each other resulting:

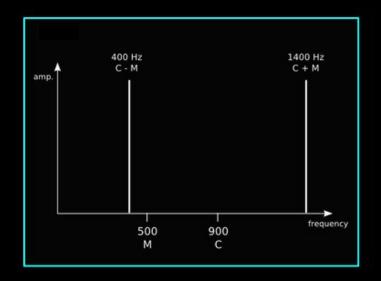
$$sin(\alpha n + \phi)sin(\beta n + \xi) = \frac{1}{2}(cos((\alpha + \beta)n + (\phi + \xi)) + cos((\alpha - \beta)n + (\phi - \xi)))$$

We obtain two partials, one at the sum of the two original frequencies and one at their difference.

This shifts the component frequencies of a sound

Doctor Who Cyberman voice





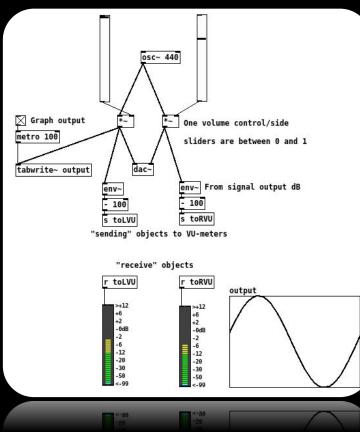
Stereo Patch

- This patch is stereo: how would you do this ?
- We want to visualize the output dB: which object?

Introducing the wireless connexion in Pd:
 → Use a "sending" object and a "receive" object as [s *] and [r *]

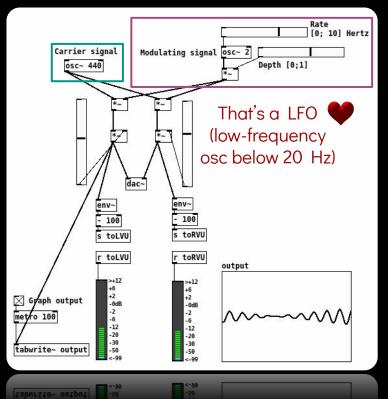
To use the VU-meter we need [env~] which take a signal and output its RMS amplitude in dB.

Stereo Oscillator

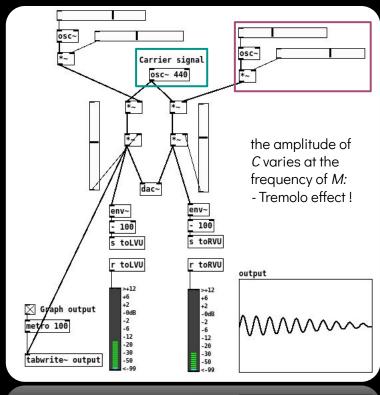


RM SYNTHESIS Using one oscillator to modulate the gain of an other one

According to your intuition, how would you patch a RM?



 Let's create a tremolo from this: adding RM on both sides



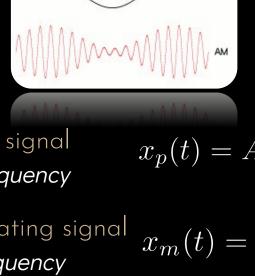
AMPLITUDE MODULATION

Definition:

Varying the amplitude of a high frequency signal, the carrier signal, as a function of a lower frequency signal, the modulating signal (commonly the one containing the information to be transmitted).



Carrier signal high frequency



Earliest method to transmit audio in radio broadcast

 $x_p(t) = A_p \cos(\omega_p t)$

Modulating signal low frequency

$$x_m(t) = A_m \cos(\omega_m t)$$

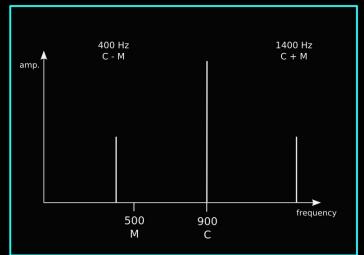
W Output:
$$y(t) = x_p(t) + k x_p(t)$$
 :

AMPLITUDE MODULATION

 $A(t) = C(t) \times (M(t) + 1) \quad \text{The modulator M is unipolar, typically set between 0 and 1.}$ $\frac{A_c A_m}{4} \left[\sin \left(2\pi (f_c - f_m)t + \frac{\pi}{2} \right) + \sin \left(2\pi (f_c + f_m)t - \frac{\pi}{2} \right) \right] + \frac{A_c}{2} \sin (2\pi f_c t)$

The carrier frequency is preserved and the sidebands generated are at *half* the amplitude of the carrier amplitude.

Having the carrier frequency, it is then possible to demodulate the signal in order to access the information hold by the carrier using a pass band.



AM WITH COMPLEX SIGNAL

Making Alien's voice with amplitude modulation Which oscillator would you choose?

Which object will we use to catch our voice?

→ [adc~]

adc~ line in from the sound	card.
mtof phasor~	
Multiply the audio si	gnal from the soundcard input by the
	es a modulation of the gain of the output when both signals are present
signal. Audio is only	
signal. Audio is only	output when both signals are present
<pre>signal. Audio is only at the input.</pre>	output when both signals are present

FREQUENCY MODULATION

Definition:

The information contained in the modulating signal is carried by varying the frequency of the carrier signal.

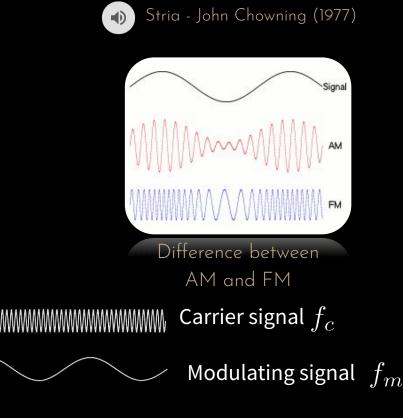
- Generally more robust than AM to transmit messages (less noise).
- Instable compare to AM regarding synthesis.
- Gives "natural" (& beautiful) sounds.

Mathematical intuition:

sinusoid modulated by another sinusoid.

With: f_c carrier frequency, f_m modulating frequency and I_m modulation index, then:

output: $y(t) = \sin(2\pi f_c t + I_m \sin(2\pi f_m t))$



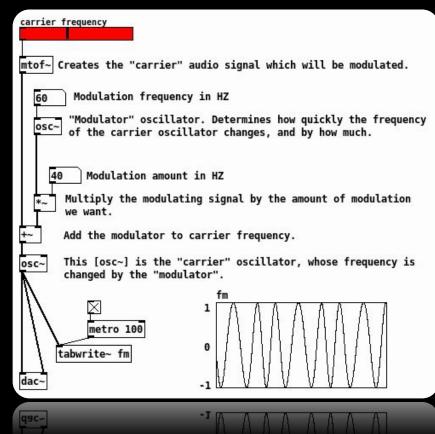
FREQUENCY MODULATION

 $y(t) = \sin(2\pi f_c t + I_m \sin(2\pi f_m t))$

You can add colors on your objects to make your patch simpler to read

- For a very small amount of modulation:
 vibrato
- For a greater amount of modulation: glissando, or sweeping.

Simple fm patch



02 Miscellaneous

Step sequencer, sound files & Kicks

The Counter

Definition of step sequencer:

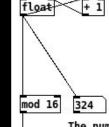
MIDI-based tool that divides a measure of music into a predetermined number of note value called steps.

We first need a counter !

How would you do it ? We want 16 steps sequencer

- You will probably need [mod *] which wrap number around given value.
- we want as output an increasing number modulo 16.

500 [metro] will rhythm your sequencer, here every 500 [metro] milliseconds.



[float] stores a Float on it's cold inlet, and outputs it when it gets the message "bang" on the hot inlet. Every time this [float] gets a bang, it sends the number stored on the left and gets a new number from [+ 1], which is one greater, to store for the next "bang".

The numbers coming from our counter will increase endlessly. We use a [mod] object to wrap these numbers around from 0 to 15

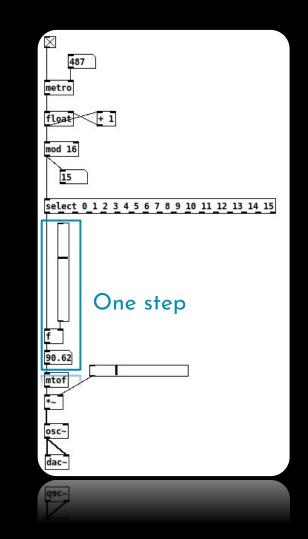
he numbers coming from our counter will increase endlessly. We use a [mod] object to wrap these numbers around from 0 to

The Step Sequencer

We will trigger a bang step by step using [select *] which compare numbers and send a bang if matching to the message.

When you are happy with one of your step, just copy paste 15 times.

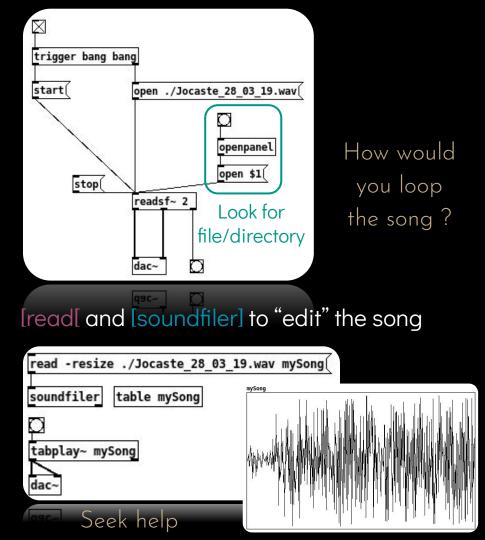
You can change osc, add enveloppes, configure your patch so that it plays harmonically, plays samples instead of notes...

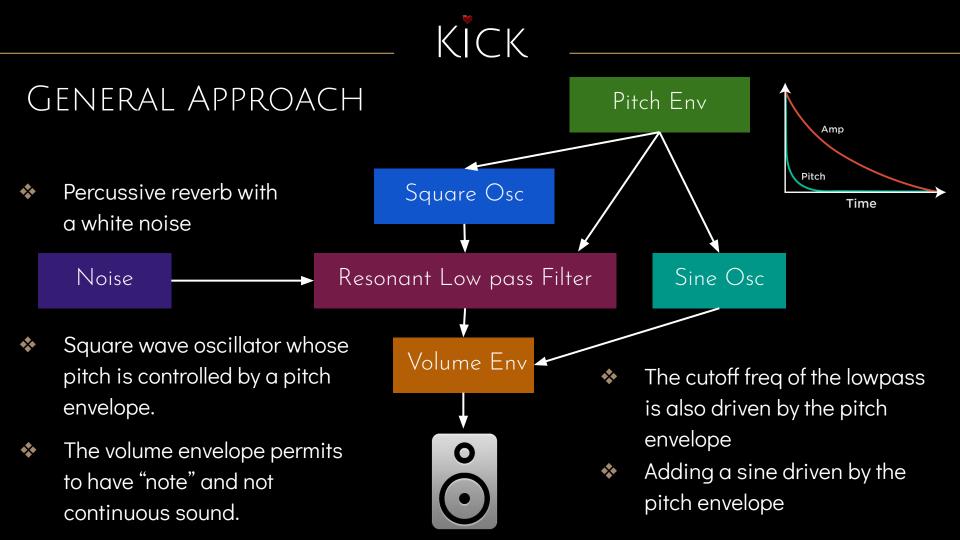


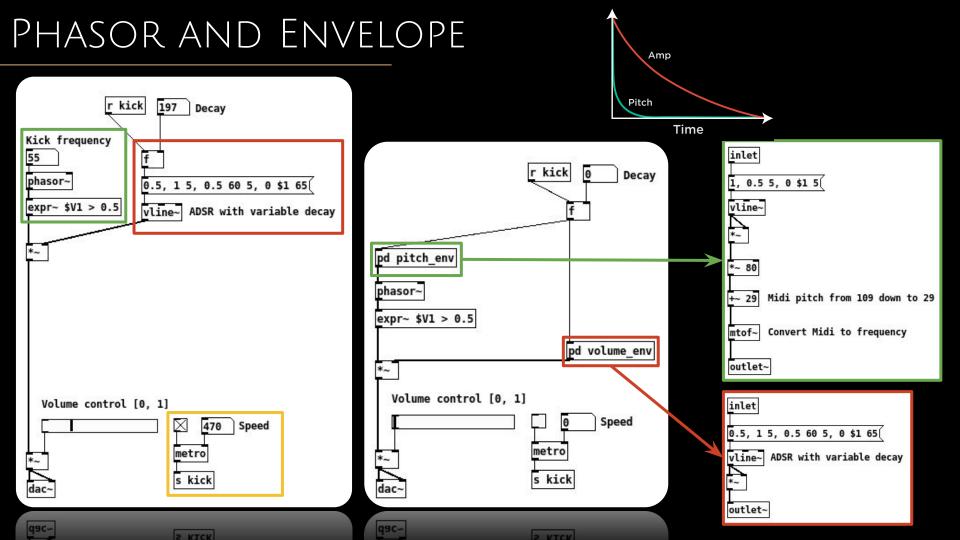
Read Files

Reminder about files formats

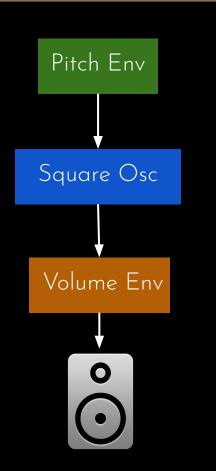
- No compression: .way or .aiff
- Compression but no quality loss: .flac
- Compression and quality loss: .mp3
- Pd objects depending on the format [readsf~] for .wav
- other formats not included in Pd Vanilla
- Don't use *space* or *special characters* for the name of your sound files
- Try to put your music in the same folder as your patch





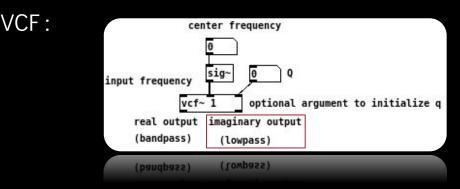


FIRST STEP ACHIEVEMENT



Next :

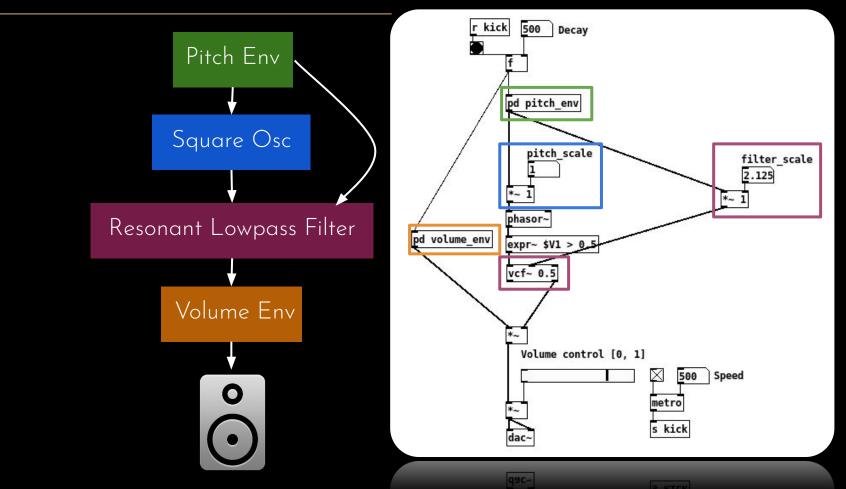
adding a resonant low-pass filter,
With the cutoff driven by our pitch enveloppe.
Which object do we need ?

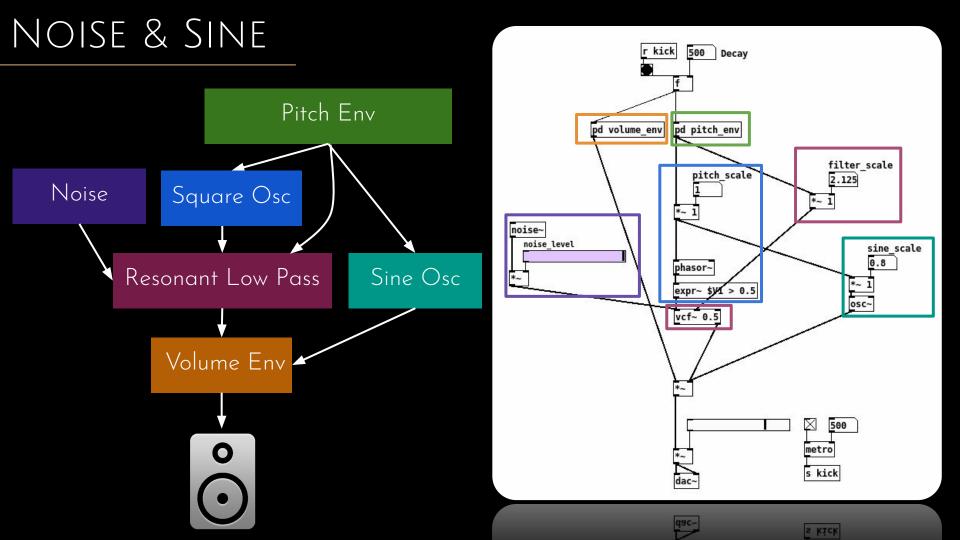


Where will we plug our pitch env?

First, let's add a pitch scale, then the vcf with a filter scale.

Resonant Low Pass Filter & Scales

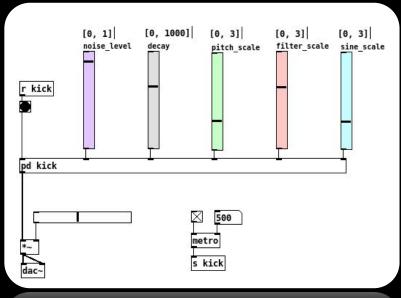


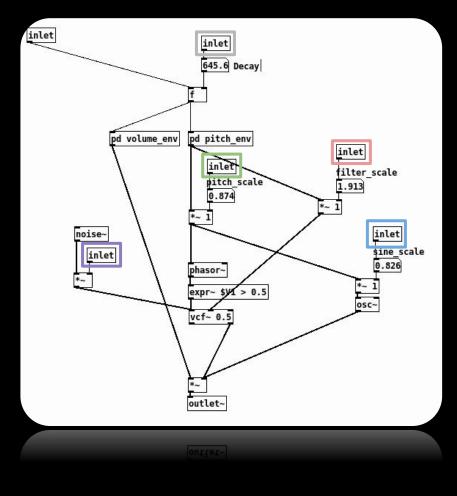


CLEAN IT UP

How to simply subpatch:

- put your inlets and outlets appropriately
- cut your boxes with them
- plug respectfully with the order

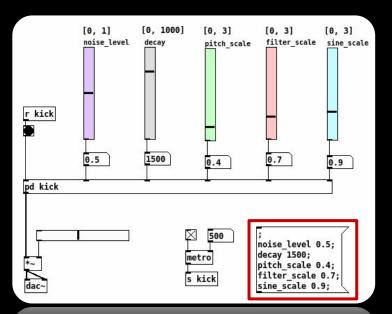




ADD PRESETS

Messages boxes to add presets:

- First add to all the concern sliders a label in "receive symbol"
- Then write a message beginning with;
 followed by all the label you want to drive

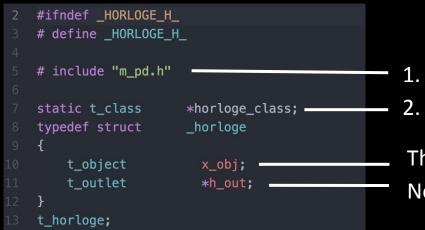


Propriétés de l'objet Slider vertical
Largeur : 15 Hauteur : 128 Plage de valeurs Inférieur : 0 Supérieur : 3
Paramètres
lin Pas d'init. Glissant
Messages
Envoyer au symbole : Recevoir du symbole : pitch_scale
Label
pitch_scale Offset en X : 0 Offset en Y : -9 DejaVu Sans Mono Taille : 10
Couleurs
Arrière plan Premier plan Label
Créer la couleur 0= =0 Label de test
Annuler Appliquer OK
Annuler Appliquer OK

O3 Externals

Create your own PD boxes

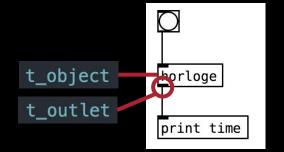
- So what is going on **inside** a given box ? So mysterious...
- We can even go deeper (and deeper ... hmm) in Pure Data objects
 - Possibility to *define your own* boxes :-) Oh woah !
 - The overall system defines **PD** externals
- PD provides a set of *includes* and *specs*
- Simple SDK with a (relatively) clear notation
- Here we will code in C (exciting hmm) but still talk about *objects*
 - Entirely dynamic linking / Runtime class loading
 - Everything defined as a C struct (erf)
 - Then simply a set of functions.



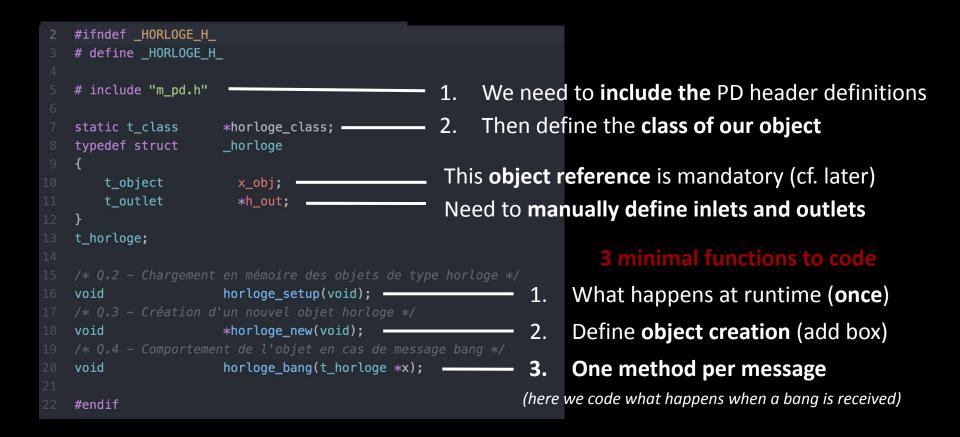
We need to include the PD header definitions
 Then define the class of our object

This **object reference** is mandatory (cf. later) Need to **manually define inlets and outlets**

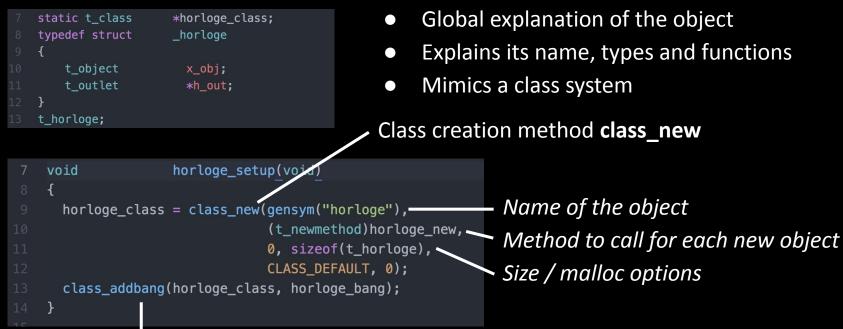
All objects have a default left-most hot inlet



Here we want to code a simple object => bang prints time



Reminder of the data structure



1. Runtime function (* setup)

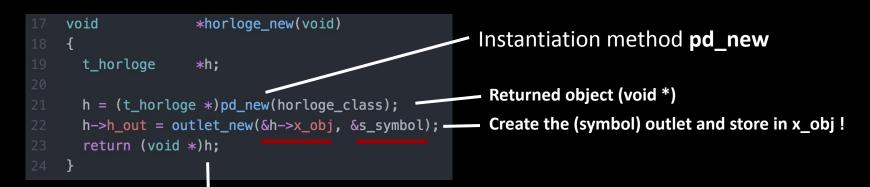
Add the behavior for bang with class_addbang Later we will also use class_addmethod (messages)

Reminder of the data structure

<pre>static t_class</pre>	<pre>*horloge_class;</pre>
typedef struct	_horloge
{	
t_object	x_obj;
t_outlet	<pre>*h_out;</pre>
}	
t_horloge;	

2. Box creation function (*_new)

- Function called when we create a box
- Similar to an object *constructor*
- Explain all initialization stuff



Return the created object

Reminder of the data structure

8 typedef struct _horloge 9 { 10 t_object x_obj;	
10 t_object x_obj;	
<pre>11 t_outlet *h_out;</pre>	
12 }	
13 t_horloge;	

3. Message handling (*_bang)

- Function called when we receive a message
- Here specific example of a *bang*
- Beware of the processing time !

Specific object instance

27 void horloge_bang(t_horloge *x)
28 {
29 time_t rawtime;
30 struct tm *timeinfo;
31
32 time(&rawtime);
33 timeinfo = localtime(&rawtime);
34 outlet_symbol(x->h_out, gensym(asctime(timeinfo)));
35 }
36 Write information to a specific outlet
37 You does not be a specific outlet
38 You does not be a specific outlet
39 You does not be a specific outlet
30 You does not be a sp

What happens for signal stuff ?

38 38		 Fonction centrale effectuant le calcul */ Fonction centrale effectuant le calcul */
	t_int	<pre>*myfft_tilde_perform(t_int *w);</pre>
	/** Q.4	– Ajout de l'objet myfft~ à l'arbre de traitement DSP */
	void	<pre>myfft_tilde_dsp(t_myfft_tilde *x, t_signal **sp);</pre>
	/** Q.3	– Libération de la mémoire de l'objet myfft~ */
	void	<pre>myfft_tilde_free(t_myfft_tilde *x);</pre>

void	<pre>myfft_tilde_setup(void)</pre>
{	
<pre>myfft_tilde_class</pre>	<pre>= class_new(gensym("myfft~"),</pre>
	<pre>(t_newmethod)myfft_tilde_new,</pre>
	<pre>0, sizeof(t_myfft_tilde),</pre>
	CLASS_DEFAULT,
	A_DEFFLOAT, 0);
class_addmethod(m	<pre>yfft_tilde_class,(t_method)myfft_tilde_dsp, gensym("dsp"), 0);</pre>
CLASS_MAINSIGNALI	N(myfft_tilde_class, t_myfft_tilde, f);
}	

- 1. Our own perform function
- 2. The DSP call (block_size dependent)
- 3. Memory liberation

Similar class setup method

Need to add the DSP function

22 void myfft_tilde_dsp(t_myfft_tilde *x, t_signal **sp)
23 {
24 dsp_add(myfft_tilde_perform, 4, x, sp[0]->s_vec, sp[1]->s_vec, sp[0]->s_n);
25 }

DSP call fills the rightful buffers