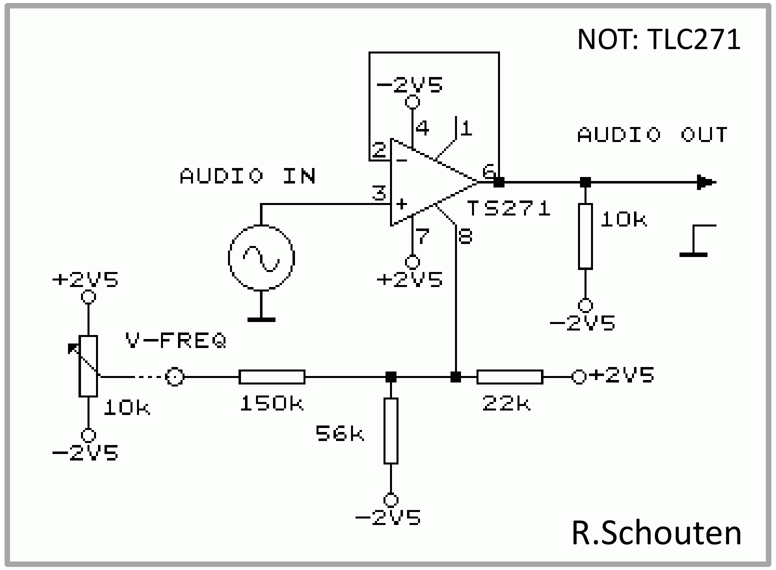
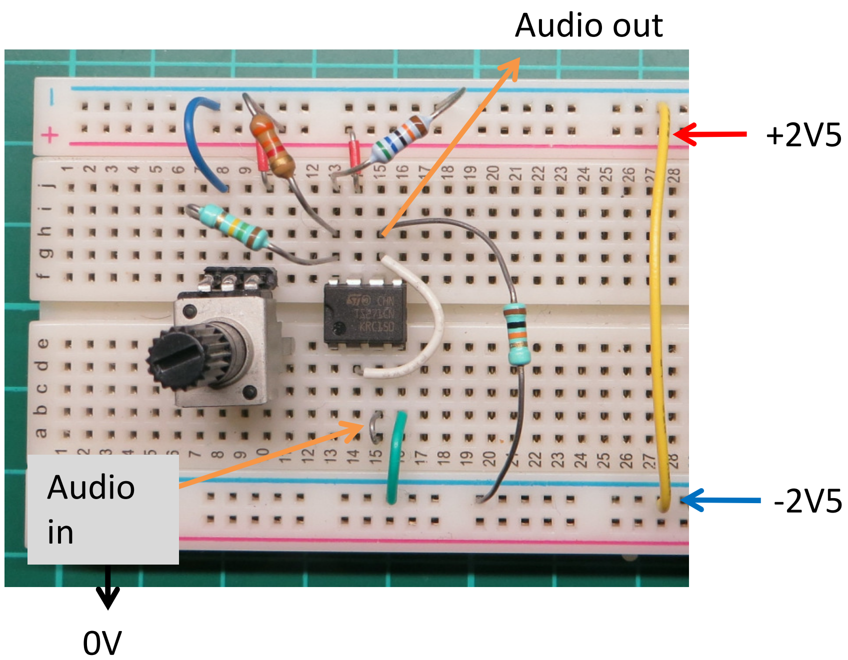
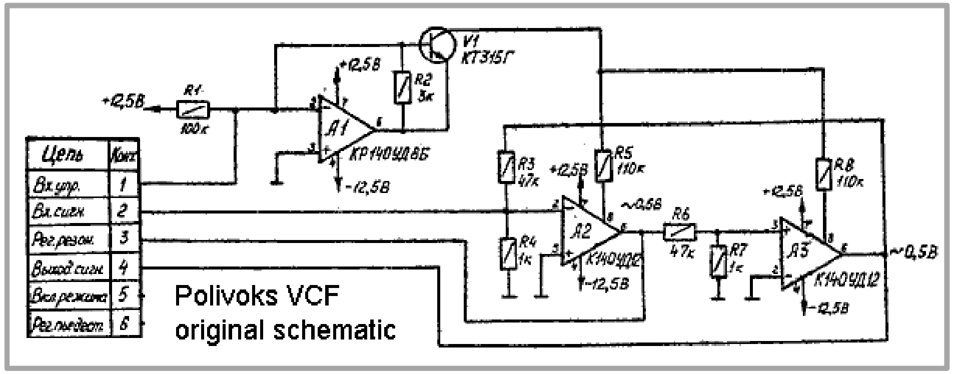
Analog synthesizer filter (lowpass) with TS271, basic concept of using supply bias for F-control



Almost every filter is based on using components like capacitors or inductors, this one is not. This one uses a specific opamp feature found on a few models that is called "adjustable supply current" (these opamps have a dedicated pin for that). By adjusting this current over a wide range the bandwidth of this opamp changes from 100kHz down to 10Hz (limited to 16KHz-30Hz in this circuit) . The 1982 Russian PoliVoks synthesizer used this trick also. The circuits described here and on next slide are completely different and simpler (using 1 available opamp instead of 3 obsolete opamps).

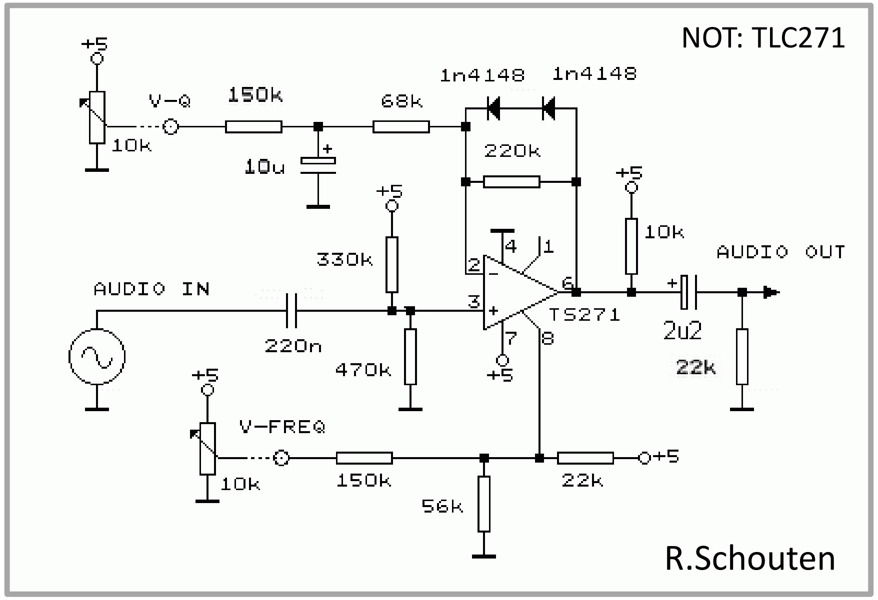


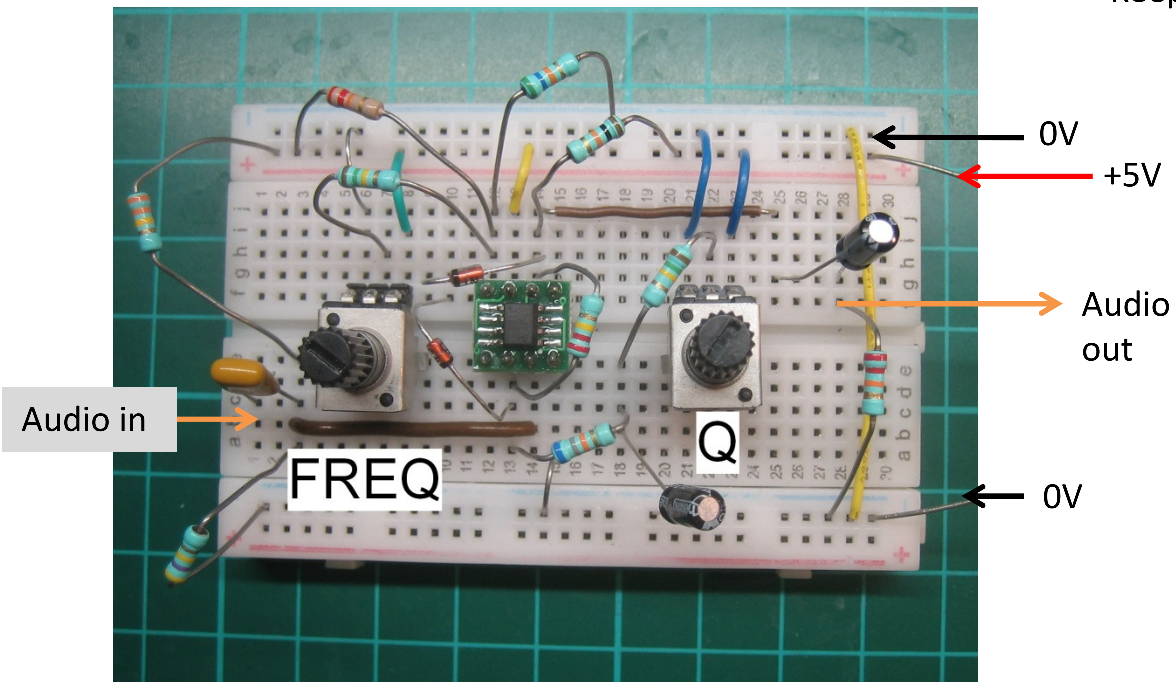


The circuit on this slide is running on bipolair supply to make it most simple, needing no coupling capacitors and dc-shift. This makes it also more clear that this filter needs no capacitors.

The circuit on the next slide runs on a single 5V supply and is therefore easy to combine with microprocessors

Analog synthesizer filter (lowpass) with TS271, final circuit with F-control and Q-control





V-Q shifts the dc-level of the output and at the point where the diodes become forward biased, the loop gain (not the circuit gain) increases. This lowers the stability and increases gainpeaking at the filter cutoff frequency up to the point where oscillation occurs. (details on last slide)

To maximize gainpeaking, the opamp has been made less stable by the 10k pull-up resistor at the output. This sets the output stage in a region with low phase margin. (See last slide)

Keep output loading > 10kohm

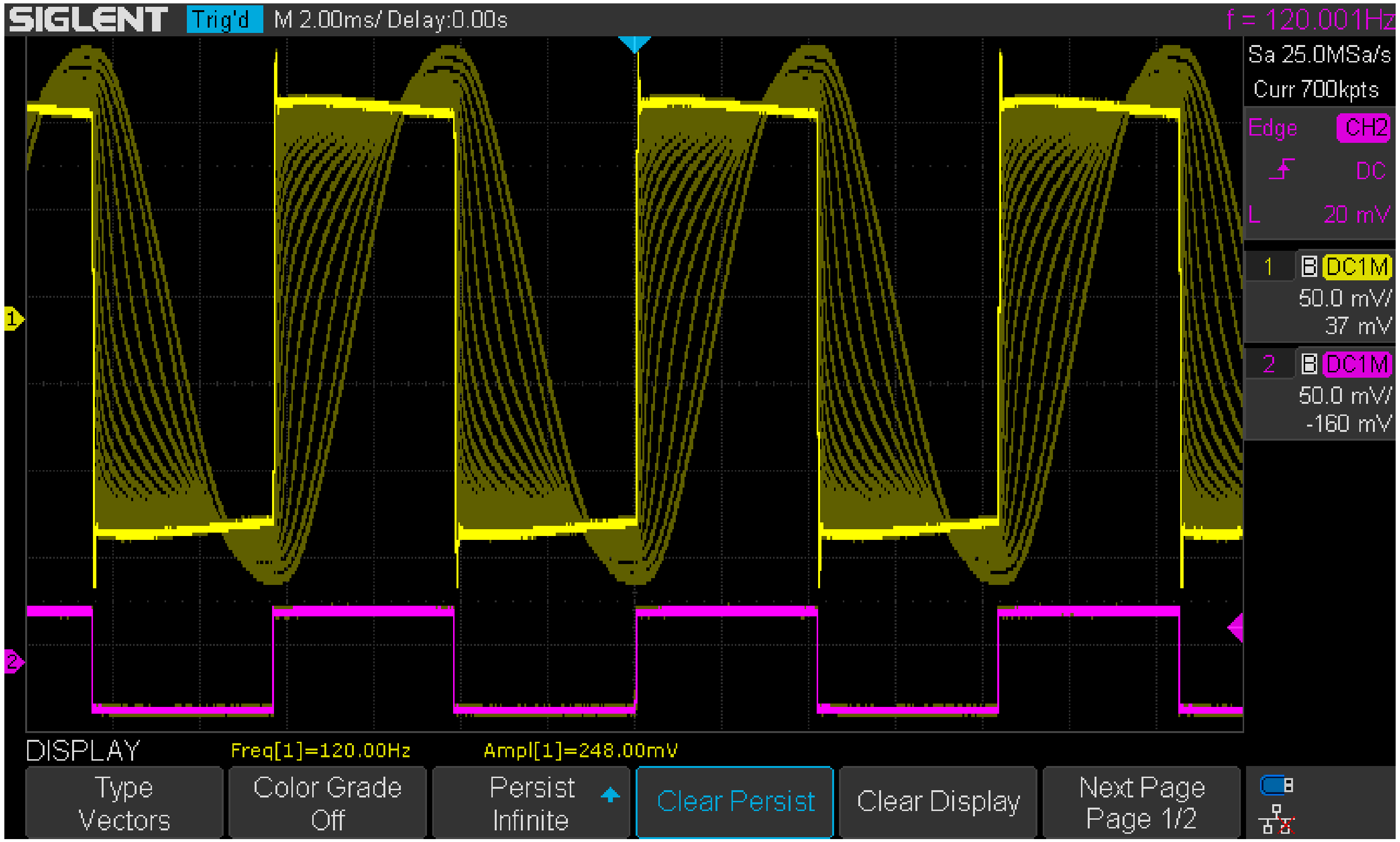
Opamp should be the TS271 from manufacturer STMicroelectronics (and not the TLC271 or likewise chips from other manufacturers).
I build over 10 circuits using different versions of this chip like SOIC packages TS271IDT (industrial) TS271CDT (commercial) and the now obsolete DIP package TS271CN. They all work fine and have quite identical performance.
I also tested the TLC271 (Texas Instruments) and that model does NOT work here, although some sellers tell you it is the same chip, it is not for this purpose.

Analog synthesizer filter (lowpass) with TS271, Frequency control sweeping:

Here the scope is set to infinite persistance (screen memory)
The filter is sweeped from a low cuttof frequency to a high one.
As a result, the output signal gradually changes from sine wave to square wave

out

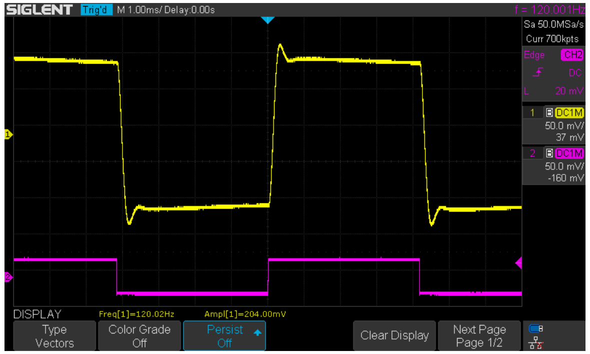
in

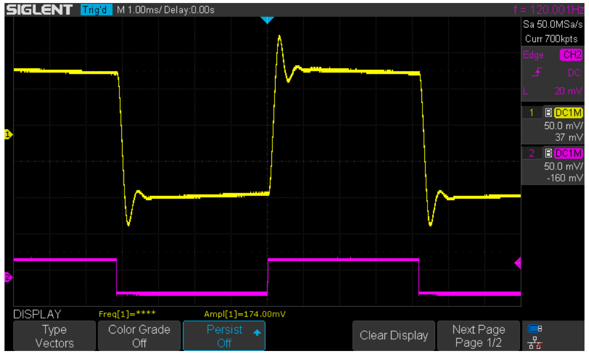


Analog synthesizer filter (lowpass) with TS271, Q control stepping:

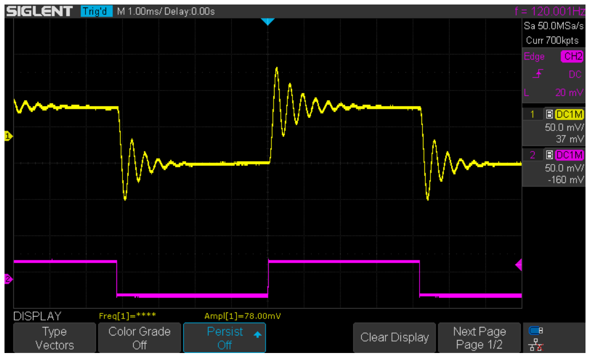
Here the filter is set to a fixed cuttof frequency approx 12x above the input frequency.
On these slides the Q is increased from lowest setting up to high (not max, see next slide).
As a result, the output signal gradually shows increased oscillations at F-cuttoff

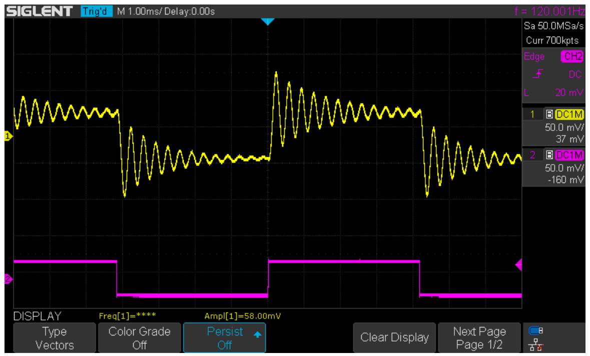
Lowest Q

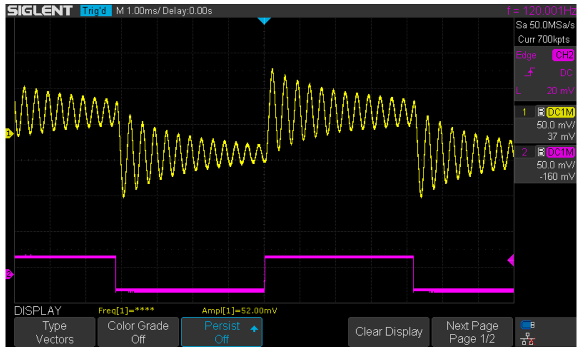




High Q



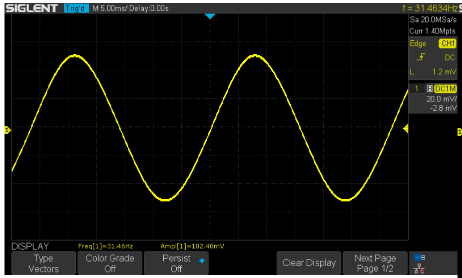




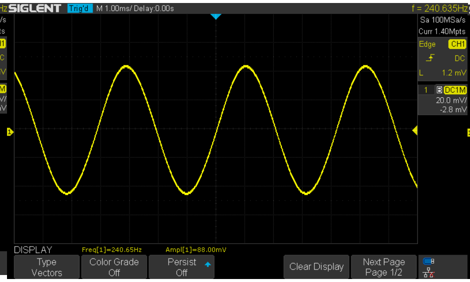
Analog synthesizer filter (lowpass) with TS271, Q set to sinewave selfoscillation:

Here the filter is set to maximum Q, resulting in self oscillation (input grounded, no signal).
The output signal is a sinewave. On these slides the frequency control is increased from 30Hz up to 15kHz
The oscillation is maintained over the whole band, the amplitude is -3dB at 10kHz.

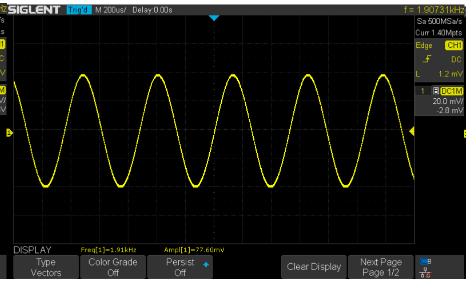
30Hz



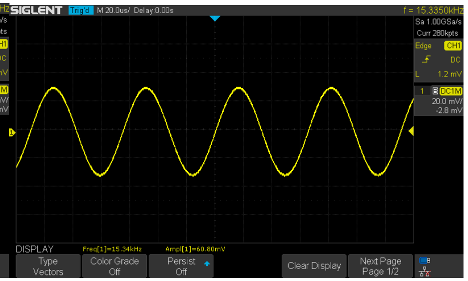
240Hz



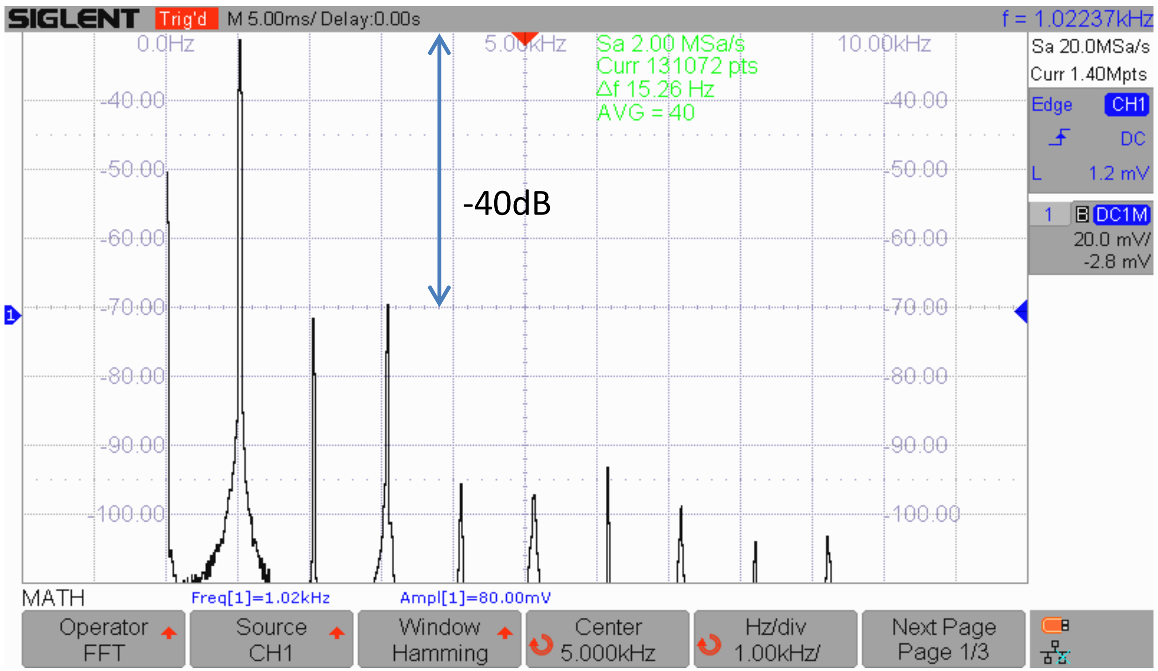
2kHz



15kHz



Spectrum from filter selfoscillation
when set to 1kHz
Shows it is quite close to a sinewave



Analog synthesizer filter (lowpass) with TS271, Frequency control has exponential curve

Musical notes follow an exponential frequency curve. Each octave is a factor of two in frequency.
For this reason it is attractive to have a filter that also follows this exponential curve for it's cutoff frequency (versus control voltage). This way the filter can track the note that is played. Another advantage of this control curve is that the controlling DAC does not need to have a high resolution, adequate resolution can now be obtained using an 8 bit DAC.

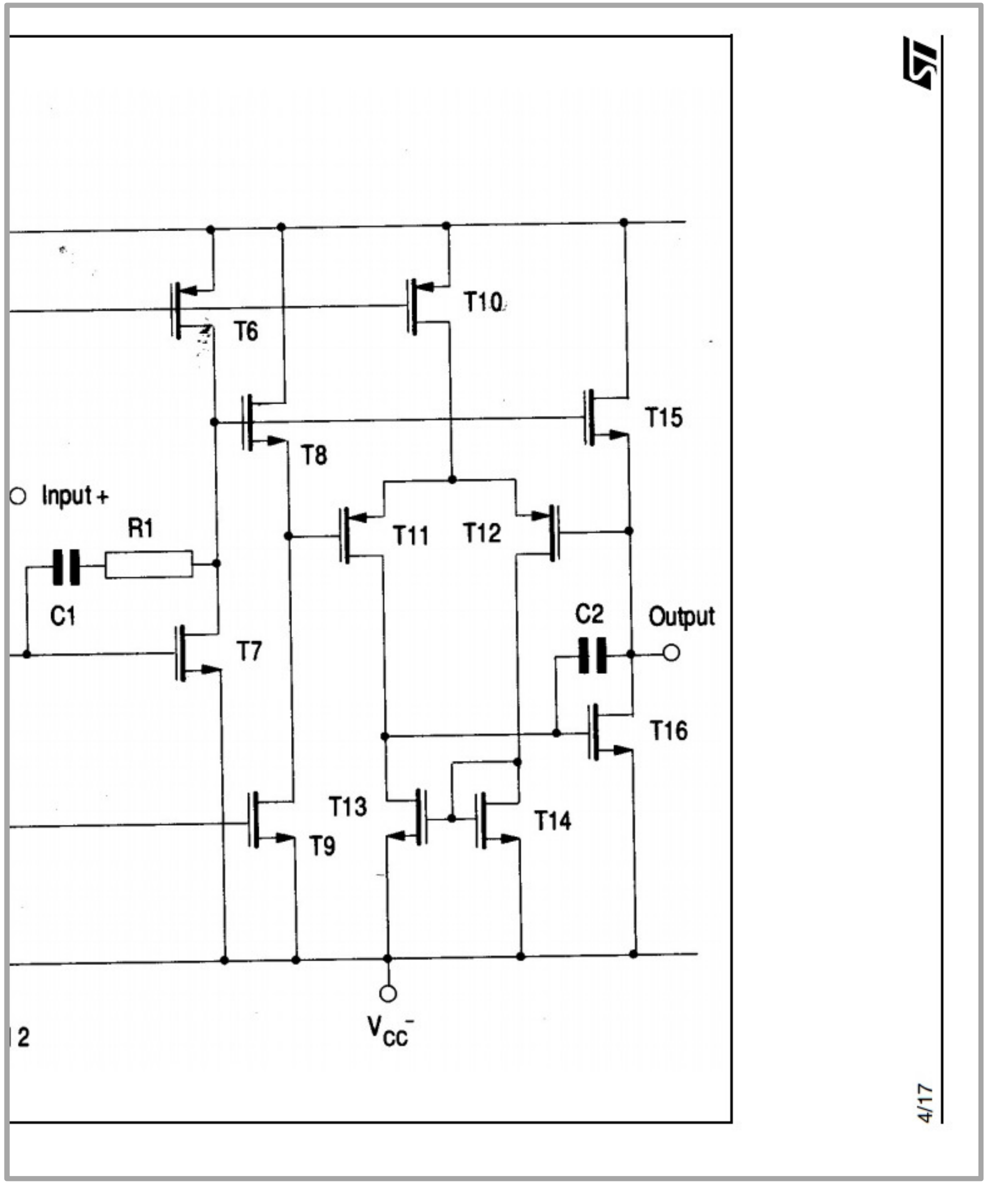
Measured control voltage that is needed to increase octaves in cuttof frequency, it shows a constant **exponential frequency versus control voltage behaviour** of ≈540mV/oct over 8 octaves.
For this measurement the filter was set to self-oscillation mode (highest Q setting)

C3 =B2-B3

	A	B	C	D
1	F-oscillation [Hz]	V-potmeter[mV]	mV/oct	Output amplitude [mVpp]
2	30	4986		100
3	60	4448	538	95
4	120	3916	532	91
5	240	3383	533	88
6	480	2853	530	84
7	960	2324	529	81
8	1920	1792	532	77
9	3840	1249	543	73
10	7680	681	568	68
11	15360	70	611	61
12				

Analog synthesizer filter (lowpass) with TS271, Q-control and chip inner structure

Inside structure of chip TS271



The datasheet shows the inners of the chip output stage. Looking into the chip, the output stage is quite asymmetric: delivering (=sourcing) current comes from a low impedance FET-source output T15, but sinking current is a done with a high impedance drain output from T16 having a feedback capacitor.

So having the output stage sinking dc current sets T16 to work and leads to more phaseshift in the feedback loop. This opamp already specs quite a low phase margin of 25deg (loopgain=1 supply=5V) which makes it sensitive to gainpeaking, an advantage for this application.

This is also related to how the voltage controlled Q works. To make the effect variable, in the circuit design 2 diodes are used as dc-current controlled variable (ac) resistor to change circuit loopgain and thereby change the damping of the filtercircuit.