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On the Analysis of Switched Continuous Time Filters.

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Abstract— **A study of the operation of switched continuous time filters (SCTF), defined as continuous time filters with elements that are alternatively switched on and off in the signal path, is conducted. A detailed calculation of the output of a SCTF in the frequency domain, which allows a fast but exact analysis of any SCTF in a general framework, is presented. Several applications of SCTFs are shown and examined using the developed theory, including a non-ideal Sample & Hold, the realization of fully integrated extremely large time constants, filter tuning by varying the duty-cycle of the switching. A detailed noise analysis for switched operated filters is also presented.**

I. SWITCHED CONTINUOUS TIME FILTERS

Consider the case of an active bandpass filter, built with operational amplifiers, capacitors, and resistors, but each capacitor has an ideal switch in series to connect/disconnect them from the filter at regular time intervals. If the capacitors are connected, the filter acts like a continuous time one; but when the switches are open the capacitors preserve their charge and the output of the filter is assumed to remain constant. The filter is not a sampled-signal one because calculations are not performed between samples. In 'active' time intervals when the passive elements are connected, the filter is continuous-time, but during 'hold' time no filtering takes place, only the state of the filter is kept in an analog memory. This filter is an example of what will be referred to a switched continuous time filter (SCTF). SCTF examples may include the switched operation of G_m-C, Mosfet-C, active or passive filters. Switched operation of filters has been studied in the past using different approximations ([1],[2]) but the theoretical background in this paper will provide a general tool, to study the behavior and limitations of any SCTF in the frequency domain.

For a linear, continuous time filter $H(f)$, and an input signal $x(t) \leftrightarrow X(f)$ (in the time and frequency domain respectively), the output signal will be $X(f) \cdot H(f)$. Fig.1 shows a scheme of a G_mC SCTF that is regularly connected during 'active' time. During "hold" time, the output and all the state variables inside the filter are kept constant. A control input *m(t)* sets the filter to "hold" or "active" depending on its value. For the sake of simplicity $m(t)$ is considered as a pulse train of frequency f_S, $T_S = f_S^{-1}$ and pulse width τ: $m(t) = \sum_{n=-\infty}^{\infty} p(t - nT_S)$. With $p(t) = 0$ if $|t| > \tau/2$, $p(t) = 1$ if $|t| < \tau/2$ thus $P(f) = \tau \cdot \text{sinc}(\tau \cdot f)$, where the *sinc* function is defined: $\text{sinc}(z) = \frac{\sin(\pi \cdot z)}{\pi \cdot z}$. Part of our objective

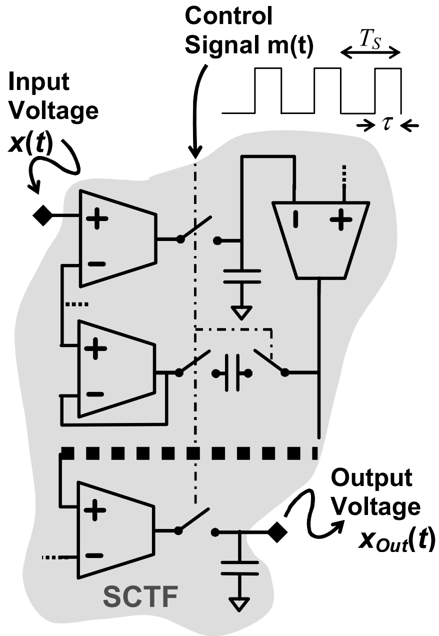

Figure 1. A switched continuous time G_mC filter.

is to calculate $x_{Out}(t) \leftrightarrow X_{Out}(f)$ in Fig.1 in terms of $X(f), H(f), T_S, \tau$. Several helpful, SCTF circuits will be also presented and examined along this work.

II. CALCULATION OF THE OUTPUT OF A SCTF IN THE FREQUENCY DOMAIN

A SCTF is not a time-invariant system so it is not possible to define a transfer function $H_{SCTF}(f)$ such that the output of the filter can be calculated as $X(f) \cdot H_{SCTF}(f)$. However it is possible to calculate the output $X_{Out}(f)$ of Fig.1 in the same way that is possible to write the output of an ideal Sample & Hold that is not a time-invariant system. Note that state variables in the SCTF are continuous in time, and they are modified only during the "active" time. Let $\mathbf{y}$ denote the vector containing the state variables of the filter. The value of the input signal during "hold" time is not relevant for the calculation of $\mathbf{y}(t)$. It is only necessary to solve the differential equations of the filter on each "active" time interval $\left(\left(nT_S - \tau/2\right) < t < \left(nT_S + \tau/2\right)\right)$, assuming as the initial condition the value of the state variables at the end of the previous "active" time interval. That is: $\mathbf{y}(nT_S - \tau/2) = \mathbf{y}((n-1)T_S + \tau/2)$. Therefore, to calculate the output it is possible to "compress" all the "active" time intervals side by side, and then solve the differential equations of the filter in a single step. Consider the input of the SCTF $x(t)$ in Fig.2(a), and the chopped signal $x_{Ch}(t) = x(t).m(t)$ in Fig.2(b). The compressed signal $x_{Comp}(t) \leftrightarrow X_{Comp}(f)$ is defined as in Fig.2(c), placing together the pieces of $x(t)$ corresponding to "active" time slots. An intermediate auxiliary function $x_I(t)$ is defined as its convolution with the impulse response $h(t)$ of the continuous time filter that is being switched:

$$x_I(t) = x_{Comp}(t) * h(t)\,. \tag{1}$$

Now $x_I(t)$ contains all the necessary information to calculate $\mathbf{y}(t)$ and the output of the filter $x_{Out}(t)$. In effect, (1) solves the filter's equations for all active times incorporating proper initial condition on each time segment. $x_{Out}(t)$ can be calculated by the inverse of the "compression" process as depicted in Fig.2(e). The output is the pieces of $x_I(t)$ during "active" time (A) and the output of the filter is assumed to be a state variable that does not change during "hold" time (B). The "compressed" signal $x_{Comp}(t)$ of Fig.2(c) where the hold times have been removed in $x_{Ch}(t)$ can be expressed as:

$$x_{Comp}(t) = \sum_{n=-\infty}^{\infty} p(t - n\tau).x(t + n(T_S - \tau)) \tag{2}$$

$$X_{Comp}(f) = \frac{\tau}{T_S}.\sum_{n=-\infty}^{\infty}\Big[\mathrm{sinc}\Big(\Big(-f.\Big(\tfrac{\tau}{T_S}-1\Big)+nf_S\Big)\tau\Big) \cdot X\Big(f.\tfrac{\tau}{T_S}-nf_S\Big)\Big] \tag{3}$$

(3) is the Fourier transform of (2). Note in (3) that aliasing may occur if the bandwidth of the input signal is larger than $f_S/2$. Also note the frequency scaling by a factor τ/T_S when evaluating the input signal spectrum. The intermediate signal $X_I(f)$ is then calculated as $X_I(f) = X_{Comp}(f).H(f)$:

$$X_I(f) = \frac{\tau}{T_S}\cdot\sum_{n=-\infty}^{\infty}\Big[\mathrm{sinc}\Big[\Big(-f\cdot\Big(\tfrac{\tau}{T_S}-1\Big)+nf_S\Big)\tau\Big] \cdot X\Big(f.\tfrac{\tau}{T_S}-nf_S\Big).H(f)\Big] \tag{4}$$

The exact output $X_{Out}(f)$ is the sum of two components:

$$x_{Out}(t) = x_{OutA}(t) + x_{OutB}(t) \quad \leftrightarrow \quad X_{Out}(f) = X_{OutA}(f) + X_{OutB}(f) \tag{5}$$

that have to be calculated separately. $x_{OutA}(t)$ corresponds to the output of the filter in the "active" time slots and is calculated by the inverse of the "compression" process in Fig.2(e). $x_{OutB}(t)$ corresponds to the output of the filter in the "hold" time slots and holds the value of $x_{OutA}(nT_S + \tau/2)$ in the last "active" time.

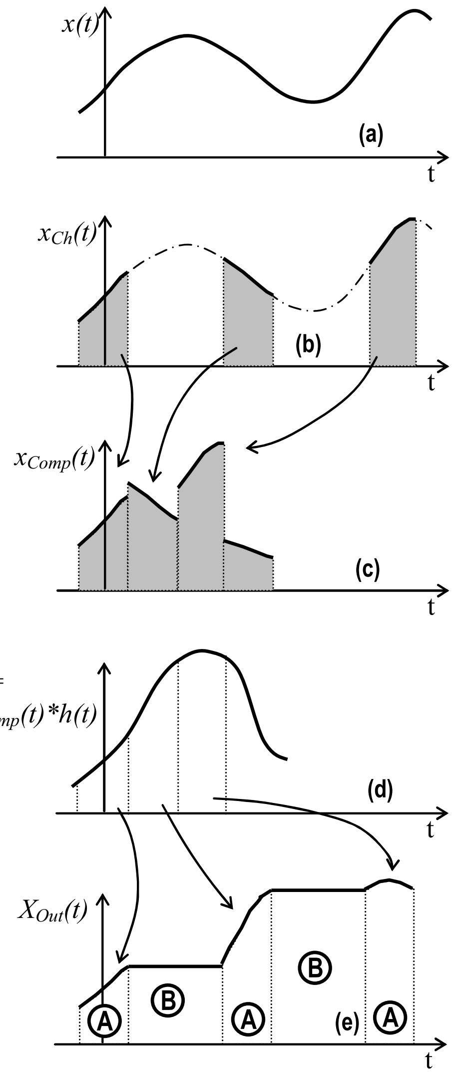

Figure 2. Evaluation process for the output of a SCTF: (a) input signal (b) chopped signal (c) compressed signal (d) intermediate signal (e) output signal for "active" time slots (A) and "hold" time slots (B).

$$x_{OutA}(t) = \sum_{n=-\infty}^{\infty} p(t - nT_S).x_I(t - n(T_S - \tau))\,. \tag{6}$$

The Fourier transform being:

$$X_{OutA}(f) = \sum_{n=-\infty}^{\infty} \mathrm{sinc}(f(\tau - T_S) + n).X_I\left(f\frac{T_S}{\tau} - \frac{n}{\tau}\right) \tag{7}$$

On the other hand $x_{OutB}(t)$ has a Fourier transform:

$$X_{OutB}(f)=\frac{T_S-\tau}{\tau}.\mathrm{sinc}\big((T_S-\tau)f\big)\sum_{n=-\infty}^{\infty}(-1)^n.X_I\left(\frac{T_S}{\tau}.(f-nf_S)\right) \tag{8}$$

Equations (4),(5),(7),(8) allow to compute the exact output signal of the SCTF in terms of the continuous time filter transfer function $H(f)$, the input signal $X(f)$, and the switching parameters T_S,τ. Note that the input signal is scaled up in frequency (4) and down (7),(8); but $H(f)$ in (4) is not scaled. Roughly in (4), is being filtered a frequency-scaled version of the input signal that is then downscaled. From another point of view, neglecting the effect of the modulating *sinc* functions and the effect of the aliasing (n=0):

$$X_{Out}(f)\approx X(f).H\left(\tfrac{T_S}{\tau}.f\right) \tag{9}$$

The filter $H(f)$ has been scaled to low frequency by a factor τ/T_S. In (9) the effect of resistor multiplication [2] in the case of a switched R-C filter can be clearly appreciated.

III. Switched Continuous Time Filter Transfer Function Definition

Although a transfer function in the sense of a linear, time-invariant system, cannot be defined for a SCTF, it is possible to work with the function $G_{SCTF}(f)$, defined as the output at frequency f of the SCTF when a pure sine-wave of unity amplitude $v_{in}(t)=\sin(2\pi ft)$ is applied at the input: $G_{SCTF}(f)=V_{out}(f).\delta(f)$. $G_{SCTF}(f)$ represents a pseudo-transfer function for the system because neglecting the effect of aliasing, the output is:

$$V_{out}(f)\approx G_{SCTF}(f)\cdot V_{in}(f) \tag{10}$$

(10) is valid if the input signal is band-limited below $f_S/2$. For a given SCTF, $G_{SCTF}(f)$ can be calculated with a computer program, adding a limited number of terms in (4),(7),(8), and using an input signal $X_{In}(f)=e^{i2\pi f}$ for $|f|<f_S/2$, $X_{In}(f)=0$ otherwise [3].

It should be pointed that switched operation of a filter does not affect stability. In effect, for an arbitrary initial perturbation, the non-switched system response is equal to $x_I(t)$ in Fig.2. So the solution of the differential equations of a system will be stable or not, even if the differential equation set is solved in a single step, or applying the time-domain partition-compression of Fig.2.

A SCTF adds some distortion to the signal in the sense that the input may be affected by *sinc* functions that are not constant with frequency. Also harmonics appear by the effect of aliasing. Both effects are considered in the above developed theory, and no extra distortion appears if switches are assumed to be ideal ones. The detailed analysis of a SCTF included in this section, may help to design a proper equalizer if necessary.

IV. A Non-Ideal Sample & Hold as a SCTF

MOS Sample & Hold (S&H) circuits are all based in the elementary analog switch and capacitor structure shown in Fig.3. During sample time, the switch is closed for a time τ so the capacitor is charged to the input voltage. The circuit can be seen as a switched R-C filter where the resistor is the on-resistance R_{ON} of the MOS switch. The usual approximation is to consider τ large enough to guarantee accurate samples every time the switch is closed. But if the time τ is too short or the time constant $R_{ON}.C$ is too large, the S&H becomes non-ideal and it cannot accurately charge the capacitor if abrupt changes of the input signal occur. But a non-ideal S&H can follow an input signal provided its bandwidth B is much less than the sampling frequency f_S. This case of an oversampling S&H ($f_S>B$) has been valuable in a micropower accelerometer signal conditioning circuit [4].

The output of the non-ideal S&H will be calculated using SCTF background. $T_S>>\tau$ is assumed for simplicity, thus only X_{OutB} has to be computed in (5). When $\tau\to 0$ it is possible to reduce (4),(8) to:

$$X_{OutB}(f)\approx\mathrm{sinc}(f.T_S)\cdot\sum_{n=-\infty}^{\infty}(-1)^n\cdot H\left(\frac{T_S}{\tau}\cdot(f-nf_S)\right)\cdot\sum_{m=-\infty}^{\infty}X(f-mf_S). \tag{11}$$

In this case (11) is evaluated using the R-C low-pass transfer function $H(f)=\dfrac{1}{1+j2\pi R_{ON}Cf}$. Fig.3 shows the measured

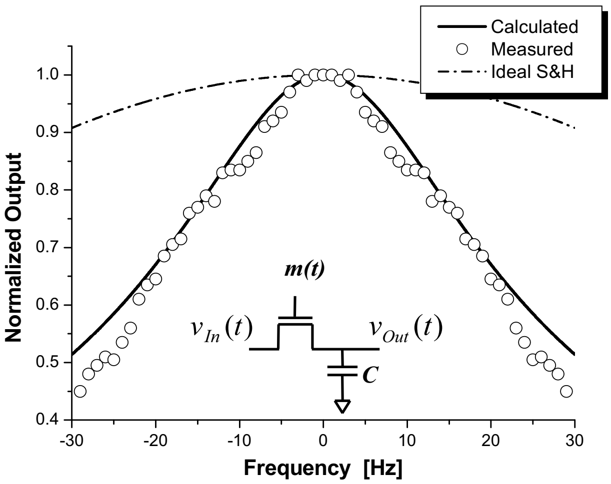

Figure 3. Magnitude of the pseudo-transfer function of a non-ideal SCTF for f_S =125Hz, τ = 0.1ms, R_{ON} = 3.9kΩ. The dashed line is classical result for the ideal S&H:

and calculated output of a non-ideal S&H built with a *W*/*L* = 100μm/5μm NMOS transistor (measured R_{ON} = 3.9kΩ), and a 100nF capacitor, switched at f_S=125Hz, τ=.1ms. The continuous line is $G_{SCTF}(f)$ as described in section III. For the non-ideal S&H, the SCTF theory allowed for the calculation of the exact distortion, which can be equalized later in the signal path (a usual practice for the ideal S&H).

Apart from the above described SCTF analysis, there are other possible approaches to examine this circuit, the simplest being traditional S&H transfer represented in the dashed line of Fig.3. Albeit simple, this approximation does not take into account non-idealities when τ is too short. Using the intuitive resistor multiplication [2], one can estimate that the transfer function of Fig.3 should be equal to a low-pass with a cut-off frequency given by T_S/τ times the *RC* constant. This second approach takes into account S&H non-ideality, however *sinc* modulating effect gets lost. An exact approach should consider the *sinc* function, small τ, and aliasing effects. In the annex at the end, an exact transfer for the non-ideal S&H was developed using the z-transform formalism. When plotted, the result is the same as (11). In spite of being derived in a more comprehensible way this result can not be easily adapted to other SCTFs like the analysis of section II.

V. VERY LARGE TIME CONSTANTS

Another interesting application of SCTFs is the realization of fully integrated extremely large time constants. The circuit in Fig.4a for example, is a bandpass G_mC corresponding to the second stage of a 0.5 - 7Hz, 40db/dec bandpass-amplifier (Gain ~ 400) for a piezoelectric accelerometer [5]. Even employing an OTA equivalent to a 10GΩ resistor, a large C_1=250pF capacitor was used in [5] to properly set the highpass pole. However in Fig.4a a switch S_1 was included to operate the feedback loop G_{m3}-C_1 as a SCTF. Switching S_1 at a 20%, or 4%, duty cycle it was possible to substitute C_1 by a 50pF, or 10pF, capacitor respectively, without a major impact in the whole transfer function of the filter as depicted in Fig.4b. Each symbol in Fig.4b represents the result of a transient simulation, for several frequencies, and C_1 value. The switch S_3 is closed in hold times, and was placed only to set the voltage at the output node of G_{m3} during 'hold'. Without S_3, it takes some time to G_{m3} after S_1 is closed, to deliver the proper current to C_1, and the amplitudes in Fig.4b result slightly different for the smaller capacitors.

VI. FILTER TUNING BY MEANS OF SWITCHING

Switched operation can be exploited, for example, to adjust the transfer function of a filter by means of the duty cycle of a digital signal. Fig.5 shows the topology of a low-pass Sallen-Key filter, including switches to operate it as a SCTF. The continuous-time transfer function $H(f)$ of the filter is the dashed line in Fig.5, and is the result of an AC analysis in SPICE simulator with both switches closed. The remaining curves and symbols in Fig.5, show simulations of the switched transfer function $G_{SCTF}(f)$ defined in (10) for different values of the duty cycle of $m(t)$. The symbols in Fig.5 were not calculated using the SCTF equations of section

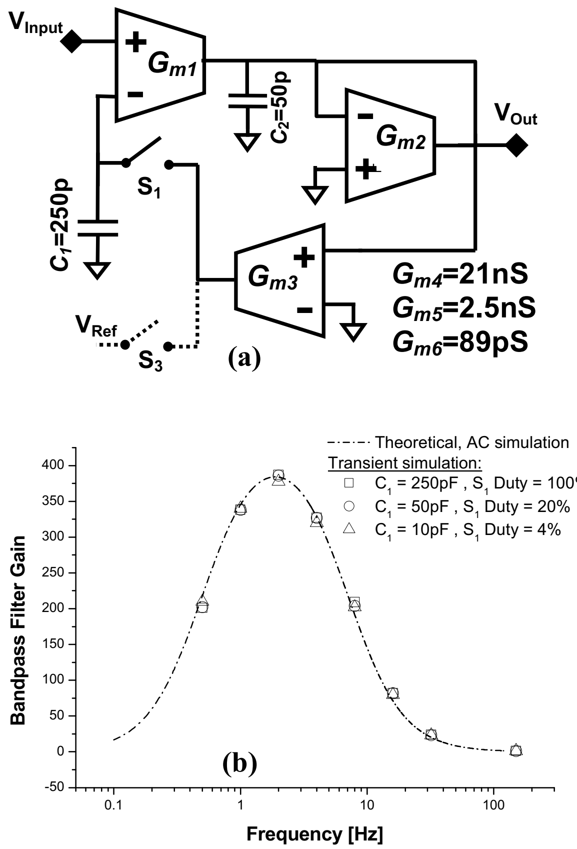

Figure 4. (a) A 2nd order bandpass G_mC, where one of the OTAs is switched to reduce the size of a 250pF capacitor to 50, or 10pF. (b) AC transfer function of the non-switched filter (line), and transient simulation for the switched version at several frequencies.

II. Instead, for each frequency and duty cycle, a SPICE transient analysis is performed. The amplitude of the output of the SCTF is measured for each simulation, and it corresponds to a single symbol in the plot of Fig.5. The operational amplifier and switches were simulated for a standard MOS 0.35μm technology. To demonstrate the accuracy of the previously developed equations, the continuous lines, which are the result of applying equations (4) to (8) to the Sallen Key filter, are shown. It should be highlighted that while each transient SPICE simulation takes a couple of minutes to complete, SCTF equations take only a few seconds to calculate the filter response over the full frequency span. For the latter also the simulation setup is simpler, just by changing a couple of equations in a MATLAB script. As predicted by (9), the filter transfer is shifted in frequency by the duty cycle of *m*(*t*).

Another example of filter tuning by adjusting the duty cycle is presented in Fig.6. Three different bandpass filters aimed for implantable medical devices, which span three orders of magnitude from few Hertz to kHz, will be implemented with the same G_mC but different switching duty cycle. These filters are a 700Hz-centered bandpass for ENG recording in [6], a 70-200Hz bandpass for cardiac activity sensing filtering [7], and the 0.5-7Hz bandpass for the accelerometer in [5]. The original filters (normalized amplitude) are ploted in Fig.6 in continuous lines. The filters

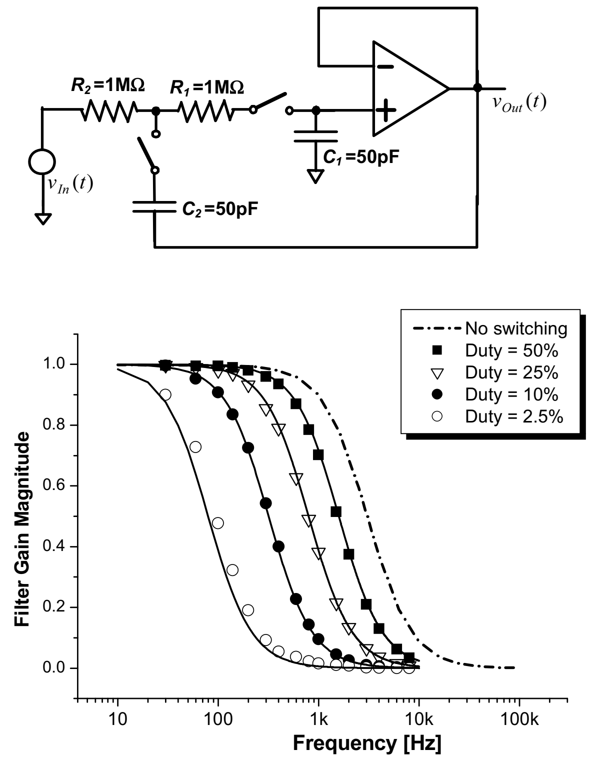

Figure 5. A switched Sallen-Key lowpass filter (top, continuous-time 3db cut-off ≅ 2kHz) and simulation (bottom) of the transfer function $G_{SCTF}(f)$ while varying the duty cycle of $m(t)$. The dashed line is the continuous time transfer function. Symbols are the result of a time domain simulation while continuous lines are obtained with SCTF theory.

were implemented with the topology of Fig.4, but an extra pair of switches were included to connect/disconect C_2. By selecting G_{m1}=20μS , G_{m2}=683nS, G_{m3}=2.2nS, C_1 = 78pF, C_2=22pF, the first transfer function is obtained (higher frequency bandpass of Fig.6). Operating with different duty cycles: 20% and 0.5%, the center frequency value can be

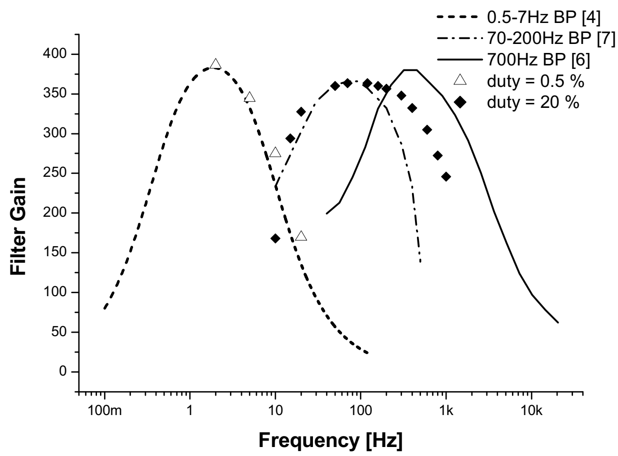

Figure 6. By switching a 700Hz-centered bandpass filter for ENG [6], a 70-200Hz cardiac sensing [7] and a 0.5-7Hz accelerometer [5] filters are simulated. Continuous lines represent original continuous time filter, while symbols represent switched G_m-C response (transient simulation).

changed to 120Hz and 2Hz, respectively. In fig 6, each symbol was obtained by a transient simulation (no lower frequency points were obtained for duty=0.5% because of convergence issues).

VII. Noise Analysis of a SCTF

Output noise in a continuous time filter is calculated by adding the noise contribution of all the elements in the filter at the output. In the case of a SCTF it is necessary to sum each noise source, but also each one passes through a SCTF to the output. Consequently noise aliasing may result in a significant contribution if the noise it is not band-limited in some way. Fortunately this limit is normally embedded in the filter. Equations (4), (7) and (8) contain all the information required for noise calculation, including the effect of aliasing.

$X_{OutA}(f), X_{OutB}(f)$ are expressed as the sums of infinite terms, in which some of these terms are also infinite sums. When calculating the output, a large enough number of terms must be summed for each infinite sum. If aliasing cannot be neglected (for example in the case of white noise), the appropriate amount of terms of $X_{OutA}(f)$ and $X_{OutB}(f)$ have to be added when numerically evaluating noise contributions with a computer. As noise is usually expressed in terms of its power spectral density (PSD), the coefficients in those equations must be squared. In this case because they are correlated, $X_{OutA}(f)$ and $X_{OutB}(f)$ cannot be calculated separately and then added, but all squared terms must be calculated together and multiplied by the noise PSD. The exact calculation can be a bit tricky, and equations (4),(7) and (8) must be all combined, and each term of the double sum must be calculated, squared and multiplied by the input before being added. To simplify calculations, a set of MATLAB routines were implemented [3] to calculate the noise contributions of a given noise source.

The thermal noise contribution of a simple G_m-C chopper amplifier [8] (Fig.8) which can be studied as a SCTF, is shown in Fig.7 and Fig.8. To compute the thermal noise contribution, $X(f)$ of section II, is substituted by a constant PSD.

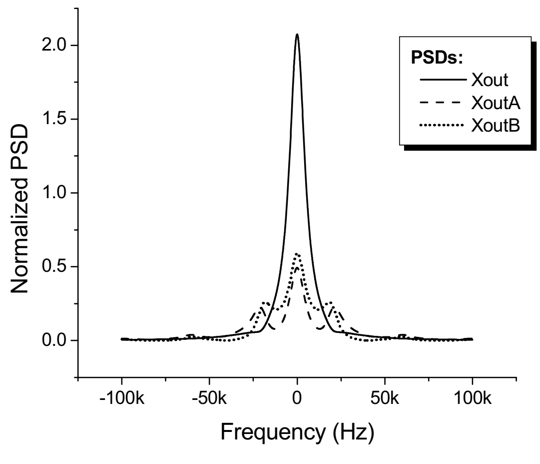

Figure 7. Normalized output noise of a chopped G_mC amplifier (continuous line); normalization is performed with respect to the white noise output at low frequency of a non-chopped G_mC. X_{OutA} (dashed) and X_{OutB} (dots) components of noise are highly correlated.

In Fig.7, the resulting normalized output noise PSD is shown for a single chopped G_m-C. In the same figure, the separate components $X_{OutA}(f)$ and $X_{OutB}(f)$ PSDs are shown. Note that $X_{OutA}(f)$ and $X_{OutB}(f)$ are highly correlated functions thus the total output PSD is not the sum of the individual ones. All results are normalized to the noise of a non-chopped amplifier. To verify the accuracy of the SCTF extension for noise contribution, Fig.8 shows a time-domain simulation of thermal noise (continuous line) compared to the PSD of X_{Out} resulting from the simulations (now dashed line). Time-domain simulation is the result of applying to the chopper amplifier a white-noise-like random input signal. Both curves were obtained with MATLAB programs, the first simulation took 15 minutes while the latter only 30 seconds to complete. Since the output of the circuit in Fig. 8 is twice that of a single G_mC, twice input noise is expected if no overhead due to switching is present, as measurements in [8] show.

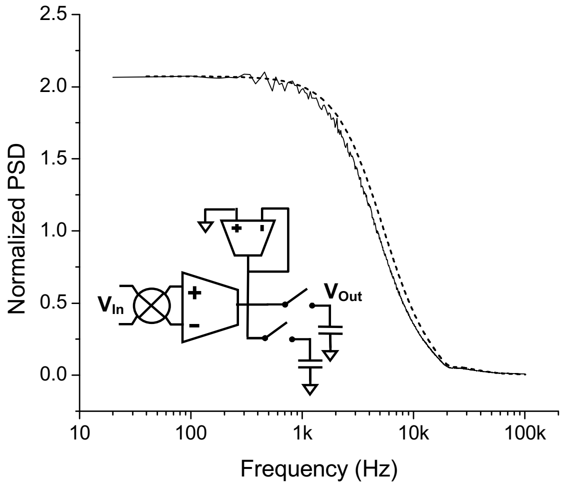

Figure 8. G_mC chopper, and its normalized output noise PSD, calculated using a time-domain simulation (continuous line) and using the proposed SCTF equations (dashed line).

VIII. Conclusion

A set of equations have been introduced to examine generic SCTFs in the frequency domain. The tool allows an exact evaluation of the output of this kind of filters, as well as to explore different design trade-offs. Particularly important is that the impact of noise, the effect of aliasing, and a transfer function definition, can be analyzed among other filter properties.

Different SCTF examples were presented: a non-ideal S&H, active filters with duty-cycle tuning, and noise analysis in a switched Gm-C chopper amplifier.

The previously developed SCTF theory allowed a better understanding of circuit operation and limitations. It should be pointed that in comparison, time-domain simulations required much more time and computer resources, and also the SCTF approach required only minimal changes on a computer program, to examine the widely different circuits studied.

IX. Annex: Alternative Calculation for the Exact Output of a Non-Ideal S&H.

The evolution of the sampled signal $x_s(t)$ is given by the following equation, where the exponential charge of the capacitor C has been introduced:

$$x_s(t) = x_s(t - T_s) + [x(t) - x_s(t - Ts)].(1 - e^{-\tau/R_{ON}C}) \qquad (12)$$

The sample time τ will be considered negligible in comparison to the sampling period T_S. Defining $\alpha = 1 - e^{-\tau/R_{ON}C}$, a discrete-time equivalent of (12) can be derived: $x_s[n] = \alpha(x[n] - x_s[n-1]) + x_s[n-1]$ which defines a discrete-time filter with input $x[n]$, and output $x_s[n]$. Its Z-transform transfer function is given by: $H(z) = \frac{X_s(z)}{X(z)} = \frac{\alpha}{1 - z(1-\alpha)}$. Calculating the frequency response of this digital filter as $H(e^{jwT_s})$, applying the usual sampled Fourier transform:

$$X_s(f) = \frac{\alpha}{1 - e^{j2\pi fT_S} \cdot (1-\alpha)} \cdot \mathrm{sinc}(f.T_S) \cdot e^{j\pi fT_S} \cdot \sum_{-\infty}^{\infty} X(f - nf_s) \quad (13)$$

Eq. (13) is an exact transfer function for the non-ideal S&H with the only assumption of infinitely-short τ. Once plotted, (13) is equal to our exact plot in Fig.3 but was derived in a more comprehensible way. However, the z-transform formalism applies only for a negligible τ, while the SCTF general formalism has no restrictions. Also, it is important to consider that the same MATLAB program was used with minor modifications to evaluate the widely different examples in sections III to VII.

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