

AN ELECTRONIC DIGITAL-TO-ANALOG CONVERTER CIRCUIT FOR A
BASEBAND TRANSMISSION SYSTEM

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5 The present invention relates to an electronic digital-to-analog converter circuit for a baseband transmission system.

10 More particularly, the invention relates to an electronic digital-to-analog converter circuit for transforming a digital signal into an analog signal without phase distortion, and in particular a circuit comprising a sigma-delta type modulator and a filter having a finite impulse response.

15 A preferred field of application of the invention is the field of mobile telephony. The invention is described essentially in the context of this field. Nevertheless, the scope of the invention should not be limited to the field of mobile telephony, but should be understood as extending to any other field in which the converter circuit and the method of conversion of the invention are applied.

20 In general, a mobile telephone or a cordless telephone can be structurally subdivided into four distinct blocks: the user block or user interface; the control block; the audio block; and the radio frequency block. The last three blocks to be mentioned constitute the radio unit of the mobile telephone.

25 Figure 1 is a block diagram showing the general structure 100 of a mobile telephone. The general structure 100 has four main blocks as mentioned above. The user of a mobile telephone has access to a user interface 120. The user interface 120 generally comprises a loudspeaker 121, a microphone 122, a keypad 123, and display means 124. The user interface 120 can optionally include other elements, for example a modem for transferring data.

35 A radio unit 130 comprises the blocks that are required for baseband transmission: speech encoding means; means for compressing information into data

blocks; and means for decompressing such data blocks into a continuous signal. These elements, and others, are shared between a control block 140, an audio block 150, and a radio frequency block 160.

5 The control block 140 comprises a microprocessor 141 which acts as a central processor unit. The micro-processor 141 performs the procedures necessary for setting up a call. It also controls the various operations of the mobile telephone by means of various
10 programs. These programs include, for example, programs for managing the user interface 120, monitoring programs (in particular for monitoring battery level), and test programs in order to facilitate maintenance of the mobile telephone. Other programs manage the connection between
15 the mobile telephone and the nearest transmission relay. The programs associated with the user interface 120 manage, in particular, interactions between the user and the other programs, in particular by interpreting the information provided by the user to the microprocessor
20 141 via the digital keypad 123 and a link 101, and by controlling the display means 124 via a link 102.

 The control block 140 also has memories 142 which are used in mobile telephones for storing, in particular, the operating system, the serial number, and the
25 telephone number associated with the mobile telephone, or indeed rights to use various services. These memories 142 can also be used when setting up a call. Data information, memory addresses, and commands are
interchanged between the memories 142 and the micro-
30 processors 141 over a bidirectional bus 103.

 The audio block 150 is essentially constituted by a signal processor unit 151 which makes use of numerous programs. The signal processor unit 151 receives
information from the microphone 122 over a link 104. A
35 link 105 serves to transmit signals between the signal processor unit 151 and the loudspeaker 121. Information is also interchanged between the signal processor unit

151 and the microprocessor 141 over a bidirectional link 106.

Communication with the radio unit 130 of the mobile telephone takes place by means of a special radio
5 frequency interface 160 in which analog-to-digital and digital-to-analog conversions are performed. The radiofrequency block 160 comprises, in particular: an antenna 161 connected to a duplexer 162; a transmitter 163; a receiver 164; and a frequency generator unit 165.
10 The microprocessor 141 manages the operation of the transmitter 163, the receiver 164, and the frequency generator 165 over respective connections 107, 108, and 109. The signal processor unit 151 can send signals to the transmitter 163 and can receive signals from the
15 receiver 164 over respective links 110 and 111. The frequency generator 165 is connected to the transmitter 163 and to the receiver 164 over respective connections 112 and 113. The duplexer 162 receives signals from the transmitter 163 over a link 114 and sends signals to the
20 receiver 164 over a link 115.

In the transmitter 163, signals carrying speech information and other information as required for telecommunication are modulated for transmission by means of a radiofrequency carrier wave. Modulation operations
25 are usually performed at an intermediate frequency which is mixed with the desired transmission frequency. Various methods of modulation are known. They depend on the type of signal and on the equipment available for transmission. To transmit analog information, it is
30 possible to use frequency modulation or frequency shift keying (FSK). To transfer digital information, it is possible to use phase shift keying (PSK) e.g. of the $\pi/4$ PSK type, or to use Gaussian minimum shift keying (GMSK).

Figure 2 is a simplified block diagram of a
35 transmission system for the transmitter 163, and more particularly of a prior art digital-to-analog converter circuit. The signal coming from the signal processor

unit 151 is sent in the form of digital data to an interface 210 of the transmitter 163. The signal received by the transmitter 163 is processed in succession by the interface 210, by a modulator 220 performing GMSK type modulation, by a digital-to-analog converter 240 (DAC), by a sample-and-hold circuit 230, and by an analog filter 250.

In the prior art, for reasons to do with energy consumption and with ease of implementation, the DAC 240 is usually a switched-capacitor DAC. Thus, for example, each group of k bits from the modulator 220 produces a voltage that is proportional to the value encoded on the k bits directly at the output from the DAC 240. That method gives rise to problems of linearity between the signal from the DAC 240 and the signal input to the DAC 240. These problems of linearity are made worse by the fact that there is a sample-and-hold circuit 230 in the above-described transmission system. The DAC 240 is a switched-capacitor digital-to-analog converter. This means that for a half-period of a cyclic clock signal controlling the transfer of data along the above-described transmission system, the values of the bits processed by the DAC 240 are not available. The sample-and-hold circuit 230 is therefore required to maintain each processed bit value for at least one half-period of the clock signal. The presence of the sample-and-hold circuit 230 nevertheless increases problems of non-linearity in the transmission of the signal. In certain transmitters, and in particular in transmitters used in GSM type mobile telephone systems, the distortion of the transmitted signal can be very detrimental to the quality of the call.

The prior art, in particular patent EP A-0642221, describes an electronic digital-to-analog converter circuit for a baseband transmission system. The circuit includes an interpolator for increasing a sampling frequency of the digital signal, a sigma-delta type

modulator, a digital-to-analog converter block comprising a finite impulse response filter (FIR), and finally an analog filter of lowpass filter type.

However, the FIR described in that document is a conventional FIR, i.e. having a transfer function of the type $h(z)=1+a_1z^{-1}+a_2z^{-2}+a_3z^{-3}+\dots$. Such a filter is not adapted to an application in the GSM field since it would require a large number of coefficients a_i resulting in an FIR of large volume and of high electricity consumption in order to obtain a filter having a steep slope.

The present invention makes it possible to mitigate the above-described drawbacks of prior art systems and, in particular, to obtain a steeply-sloped filter that is smaller than that of digital-to-analog converters of the prior art.

The present invention provides an electronic digital-to-analog converter circuit for a baseband transmission system to transform a digital signal into an analog signal without phase distortion, the system comprising:

- a digital modulator delivering a signal in the form of groups of n bits;
 - an interpolator for increasing a sampling frequency of the digital signal;
 - a sigma-delta type modulator for transforming an n -bit signal into a 1-bit signal;
 - a digital-to-analog converter block, specifically for filtering out quantization noise in sharp manner, which block comprises a digital-to-analog converter and a finite impulse response filter; and
 - an analog filter of lowpass filter type,
- the electronic circuit being characterized in that the finite impulse response filter is a multiband filter.

Various aspects and advantages of the invention appear more clearly from the following description given with reference to the accompanying figures which are

given purely by way of non-limiting indication of the invention, and in which:

- Figure 1, described above, is a block diagram shown the general structure of a mobile telephone;

5 • Figure 2, described above, is a block diagram of a prior art electronic digital-to-analog converter circuit;

- Figure 3 is a block diagram of an electronic circuit of the invention;

10 • Figure 4 is a block diagram of a sigma-delta modulator; and

- Figure 5 shows the frequency spectra of the signal as conveyed at various points in the electronic circuit of the invention while a signal is passing therethrough.

15 In Figure 3, there can be seen the interface 210 which receives the signal from the signal processor unit 151 and which is already present in the prior art transmission system shown in Figure 2. There can also be seen the GMSK type modulator 220. The signal from the modulator 220 is processed in succession by the following
20 elements: an interpolator 310; a sigma-delta type modulator 320; a digital-to-analog converter block 330; and finally an analog filter 340. The function of each of these elements is described in greater detail below.

25 The modulator 320 is known as a sigma-delta modulator. Figure 4 is a block diagram of such a modulator. The term "sigma-delta" comes from the configuration of the modulator which makes use firstly of an adder 410, as conventionally symbolized by the Greek letter "sigma". The output from the adder 410 is
30 connected to an integrator 420 which integrates the signal. The integrator 420 acts as a lowpass filter for the payload input signal. The integrator 420 is followed by a quantizer 430 operating on the principle of performing quantization as the result of taking the
35 difference between two consecutive samples of the processed digital signal. It is from this difference that the term "delta" arises, in line with the common

name for such an operation. A negative feedback loop 440 acts as a highpass filter for quantization noise as generated by the quantizer 430. The sigma-delta type modulator makes it possible to reject quantization noise into high frequencies while the payload signal is present in the low frequencies.

In digital-to-analog converter systems comprising a sigma-delta type modulator, the sampling frequency is initially increased. In this case, this operation is performed by the interpolator 310. The interpolator makes it possible to perform an oversampling operation.

In a preferred application of the invention, the frequency at which the data from the GSMK type modulator is sampled is doubled by performing linear type interpolation: between two consecutive data groups from the GSMK type modulator 220, an additional data group is added corresponding to the mean of two data groups that are directly adjacent in time. The sampling frequency is then tripled at the input to the sigma-delta type modulator by repeating each data group three times.

The interpolator 310 thus makes it possible to go from an n -bit signal sampled at a frequency F_s to an n -bit signal sampled at a frequency $N \times F_s$ where N is a natural integer. In a preferred application $N=6$ and $n=8$.

The sigma-delta type modulator 320 associated with the interpolator 310 makes it possible to go from an n -bit signal sampled at the frequency $N \times F_s$ to a 1-bit signal sampled at the frequency $N \times F_s$.

The 1-bit signal delivered by the sigma-delta modulator 320 can be a signal that is pulse duration modulated (PDM). It is supplied to the input of the digital-to-analog block. The output signal from the sigma-delta modulator 320 thus supplies the digital-to-analog converter block 330 with less information since the number of bits has been reduced, however it does so at a frequency that is higher because of the oversampling performed by the interpolator 310.

The quantization noise generated at the output from the sigma-delta type modulator is large since the signal is now coded on a single bit. The digital-to-analog converter block 330 serves to filter this quantization noise. The digital-to-analog converter block 330 is constituted by a 1-bit digital-to-analog converter, or 1-bit DAC, and by a filter having finite impulse response.

The output from the 1-bit DAC activates the finite impulse response filter that follows it. The FIR has current sources. Each bit input to the 1-bit DAC controls a current source after processing that makes use in particular of shift registers. Each current source is associated with a respective coefficient of the finite impulse response filter. Each current source delivers current that is proportional to a corresponding one of the coefficients of the FIR.

By using a 1-bit DAC as in the case of the present invention, it is possible to solve problems of non-linearity associated with the signal being transmitted through the digital-to-analog conversion system.

However, in order to filter out the high level of quantization noise, the finite impulse response filter must have a cutoff frequency close to the baseband frequency (typically 100 kHz for use in a GSM context). This means that the graphical representation of the transfer function of the FIR filter must have a slope that is very steep around 100 kHz. Unfortunately, this can be achieved only by using a conventional FIR filter that is complex, and consequently that has a large number of coefficients. This would increase the number of current sources required. Such a solution would increase consumption of electrical energy during signal transmission, and would occupy a larger volume.

In the invention, the finite impulse response filter is a multiband filter. An advantage of the FIR is that of eliminating all frequencies close to the signal without deteriorating phase, as shown by the 4th spectrum

in Figure 5 where the quantification noise 530 is filtered for the frequencies adjacent to the signal, while also having the advantage of being much smaller than a conventional FIR.

5 In addition, a multiband FIR filter is a filter that is easy to implement because its number of coefficients is small, and consequently it requires only a small number of current sources. It presents the characteristics of having a transfer function whose
10 graphical representation has a steep slope at a low frequency, while allowing higher frequencies to pass in a manner that is periodic.

 The higher frequencies which are not filtered by the multiband FIR filter are filtered by means of a simple
15 lowpass filter, such as the analog filter 340 of Figure 3.

 The presence of the multiband FIR filter also makes it possible to avoid having a complex analog filter in the signal transmission system, given that such a filter
20 would be liable to introduce phase distortion.

 Figure 5 shows the frequency spectra of the transmitted signal at various points in the electronic circuit of the invention.

 On inspecting Figure 5 from top to bottom, the five
25 frequency spectra shown are those observed respectively: at the input to the interpolator 210; at the output from the interpolator 210; at the output from the sigma-delta modulator 320; at the output from the digital-to-analog converter block 330; and at the output from the analog
30 filter 340.

 It can clearly be seen from the frequency spectrum of the signal output by the sigma-delta modulator 320, that the frequency spectrum 510 representing quantization noise is rejected into frequencies that are high relative
35 to the frequency spectrum 520 of the payload signal. In the frequency spectrum of the signal output by the digital-to-analog converter block, it can be seen that

the multiband FIR filter of the invention allows frequency bands 530 to pass through, and that consequently quantization noise is not filtered out completely, however all the frequencies thereof that are
 5 immediately adjacent to the frequencies of the payload signal are indeed filtered out.

The following analog filter 340 can thus be a filter that is easy to implement. The electronic circuit of the invention is economical in terms of volume required for
 10 physical implementation.

In the prior art, compensation circuits are used after the signal has been transformed into an analog signal. The presence of compensation circuits in the digital portion would require bits to be added to the
 15 signal to be processed, and that would consequently exacerbate problems of linearity in terms of signal transmission.

In a preferred application of the invention, the sigma-delta type modulator has a transfer function $h_1(z) =$
 20 $1 - 2z^{-1} + z^{-2}$.

In a preferred application of the invention, the transfer function of the FIR filter is:

$$h_2(z) = -1 + z^{-14} + 2.74z^{-21} + 4.85z^{-28} + 5.76z^{-35} + 4.85z^{-42} + 2.74z^{-49} + z^{-57} - z^{-71}.$$

25 Under such circumstances, the FIR filter has only nine coefficients. Consequently it is clear that it is simple to implement. Such a filter also has the advantage of being economical in terms of the volume required for being physically implemented.



CLAIMS

- 1/ An electronic digital-to-analog converter circuit for a baseband transmission system to transform a digital signal into an analog signal without phase distortion, the system comprising:
- a digital modulator (220) delivering a signal in the form of groups of n bits;
 - an interpolator (310) for increasing a sampling frequency (F_s) of the digital signal;
 - a sigma-delta type modulator (320) for transforming an n -bit signal into a 1-bit signal;
 - a digital-to-analog converter block (330), specifically for filtering out quantization noise in sharp manner, which block comprises a digital-to-analog converter and a finite impulse response filter; and
 - an analog filter (340) of lowpass filter type, the electronic circuit being characterized in that the finite impulse response filter is a multiband filter.
- 2/ An electronic circuit according to claim 1, characterized in that the digital-to-analog converter is a 1-bit digital-to-analog converter delivering a pulse duration modulated signal.
- 3/ An electronic circuit according to claim 1 or 2, characterized in that the finite impulse response filter comprises current sources delivering currents that are proportional to coefficients defining the filter.
- 4/ An electronic circuit according to any one of claims 1 to 3, characterized in that the sigma-delta type modulator (320) has the following transfer function: $h_1(z) = 1 - 2z^{-1} + z^{-2}$.
- 5/ An electronic circuit according to any one of claims 2 to 4, characterized in that the finite impulse response filter has the following transfer function:



$h_2(z) =$

$$-1 + z^{-14} + 2.74z^{-21} + 4.85z^{-28} + 5.76z^{-35} + 4.85z^{-42} + 2.74z^{-49} + z^{-57} - z^{-71}.$$

6/ An electronic circuit according to any one of claims 1
5 to 5, characterized in that the interpolator (310)
multiplies the sampling frequency (Fs) by six.

7/ The use of the digital-to-analog converter circuit
according to any preceding claim in a mobile telephone.