## 1. Design parameters

| parameter | min. | typ. | max. | unit | comment |
|---|---|---|---|---|---|
| $f_c$ | 87.5 | | 108.5 | MHz | broadcast carrier frequency |
| $\Delta f$ | | | 75 | kHz | frequency deviation |
| $B$ | | 210 | | kHz | base-band bandwidth |
| $\tau$ | | 50 | | µs | de-emphasis time constant (Europe) |
| $f_m$ | 0.03 | | 15 | kHz | modulation frequency |
| $f_s$ | | 240 | | MHz | sampling frequency |
| $f_b$ | | 960 | | kHz | base-band clock rate |
| $f_{DAC}$ | 32 | | | kHz | output clock rate for DAC |
| $f_{MCLK}$ | | | 18.5 | MHz | WM8731 master clock frequency |
| $f_{BCLK}$ | | | 20 | MHz | WM8731 bit clock frequency |

The following constraints have to be fullfilled.

$$f_s > 2 \cdot f_c$$

$$B > 2 \cdot (\Delta f + 2 \cdot f_m) \quad \text{Carson's bandwidth rule}$$

$$f_b > B$$

$$f_{DAC} > 2 \cdot f_m$$

$$f_{BCLK} \geq 2 \cdot 16 \cdot f_{DAC} \quad \text{with 16 bit audio data}$$

## 2. Downsampling to I/Q base-band

The I/Q modulator works with 1-bit signals. The upper two bits of the DDS's phase accumulator represent the quadrant.

The multiply operation is performed by a simple XOR logic. As a result, the I/Q base-band signal are triangular waves instead of pure sines/cosines which will lead to harmonic distortions of $4 \cdot \Delta f$.

## 3. Base-band filter

The base-band filter is compromised of two third order cascaded integrator-comb (CIC) filters. After the first filter the data rate is 48 MHz, after the second filter 960 kHz. The first zero is $f_c/R$ where $R$ is the decimation factor of the CIC filter.

The base-band bandwith should be larger than $B$. For optimal channel separation it should be less than 400 kHz.

## 4. Frequency discriminator

The I/Q base-band signal is connected to the CORDIC module which will calculate the phase.

$$\theta = \arctan \frac{Q}{I}$$

After differentiation of the phase the signal is restored.

$$\omega_m = \frac{d\theta}{dt}$$

## 5. Downsampling to audio sample rate

A third order CIC decimator converts from base-band sample rate to audio sample rate. This effectively removes any frequencies above 15 kHz, e.g. MPX and RDS signals. Because no sinc-correction has been made yet, the transfer function will drop at higher audio frequencies.

## 6. De-emphasis

### 6.1. De-emphasis with IIR filter

A de-emphasis filter with the transfer function

$$H(s) = \frac{1}{1 + \tau \cdot s}$$

corrects the pre-emphased audio signal of the broadcast station.

Impulse invariance method converts $H(s)$ to $H(z)$.

$$H(z) = \frac{b}{1 + a \cdot z^{-1}}$$

$$a = -e^{-\frac{1}{\tau \cdot f_{DAC}}}$$

$$b = 1 + a$$

### 6.2. De-emphasis with build-in filter of the WM8731

The DAC filter of the WM8731 can apply digital de-emphasis. The analogue transfer function is

$$H(s) = \frac{1 + \tau_2 \cdot s}{1 + \tau_1 \cdot s}$$

with $\tau_1 = 50\ \mu s$ and $\tau_2 = 15\ \mu s$. The zero at $10.6\ kHz$ partly compensates the *sinc* drop of the CIC decimation filter.

## 7. Audio interface

The Altera DE1 board contains the audio CODEC WM8731 from Wolfson Microelectronics.

After reset the CODEC will be initialized via I²C two-wire interface to 12 MHz MCLK, 32 kHz sampling rate, 16 bit audio data length, DSP/PCM mode B and enabled de-emphasis.

| register | name | data | comment |
|---|---|---|---|
| R0 | Left Line In | 010010111 | default |
| R1 | Right Line In | 010010111 | default |
| R2 | Left Headphone Out | 001111001 | LHPVOL = 0 dB |
| R3 | Right Headphone Out | 001111001 | RHPVOL = 0 dB |
| R4 | Analogue Audio Path Control | 000010010 | DACSEL, MUTEMIC |
| R5 | Digital Audio Path Control | 000000010 | DEEMP = 32 kHz |
| R6 | Power Down Control | 001100111 | POWEROFF = 0, OUTPD = 0, DACPD = 0 |
| R7 | Digital Audio Interface Format | 000000011 | IWL = 16 bits, FORMAT = DSP mode |
| R8 | Sampling Control | 000011001 | 32 kHz, USB mode |
| R9 | Active Control | 000000001 | ACTIVE |
| R15 | Reset | — | reset, when written with zeros |

Serial audio data is transferred with the signals BCLK, DACDAT and DACLRC.