1. Design parameters

parameter	min.	typ.	max.	unit	comment
f_c	87.5		108.5	MHz	broadcast carrier frequency
Δf			75	kHz	frequency deviation
B		210		kHz	base-band bandwidth
τ		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

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 $H(z)$.

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

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

$$b = 1 + a$$

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 and DSP/PCM mode B.

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
R5	Digital Audio Path Control	000000010	DEEMP = 32 kHz
R6	Power Down Control	000010111	POWEROFF = 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.