1. Design parameters

parameter	min.	typ.	max.	unit	comment
f_c	87.5		108.5	MHz	broadcast carrier frequency
Δf			75	kHz	frequency deviation
В		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.

$$f_s > 2 \cdot f_c$$
 $B > 2 \cdot (\Delta f + 2 \cdot f_m)$ Carson's bandwidth rule $f_b > B$ $f_{DAC} > 2 \cdot f_m$ 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).

$$H(z) = \frac{b_0}{1 + a_1 \cdot z^{-1}}$$

$$a_1 = -e^{-\frac{1}{\tau \cdot f_{DAC}}}$$

$$b_0 = 1 + a_1$$

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 caused by 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.

8. Antenna connection

For the antenna input an I/O input buffer without hysteresis is needed. A differential input of type LVECL at EXT_CLOCK/SW[2] was choosen.

The antenna (a wire of 1 m length) was directly connected to EXT_CLOCK. No LC-circuit for tuning was needed.

9. Results

Three local broadcast stations could be received. The audio signal was of reasonable quality. If there were harmonic distorsions they were not very audible.