Introduction

U1307 is a middle and high end dual mode mobile phone researched and developed independently by HUAWEI based on the 6245 chipset produced by QUALCOMM Incorporated. It supports WCDMA，GSM/GPRS frequency bands and Bluetooth module. It is designed in compliance with the FCC Rules and Regulations **Part.24E and Part 22H.**

Intended use statements

U1307 is a WCDMA/GPRS/GSM/EDGE Mobile Phone. It can only be work in the networks which it supports the UMTS and GSM technique. The WCDMA frequency band is Band I and Band V, The GSM/GPRS frequency band includes GSM850，E-GSM900 and DCS1800 and PCS1900, so GSM850 PCS 1900MHz and band V test data included in all the documents. If there are no corresponding networks, the RF module of U1307 will not work and no any unwanted emission will be produced.

Types of Emission

For this mobile phone, the emission designators are 300kGXW (GSM/GPRS), 300KG7W (EDGE) and 5M00F9W (WCDMA). HSDPA capability is not implemented.

Frequency Range

PCS/GPRS/EDGE1900MHz: 1850 to 1910MHz/ 1930 to 1990MHz   (Up/down stream)
GSM/GPRS/EDGE 850MHz: 824 MHz to 849MHz/869 MHz to 894MHz (Up/down stream)
WCDMA 850 MHz:: 824 MHz to 849 MHz/ 869 MHz to 894MHz (Up/down stream)

Range of Operating Power

PCS1900: 0dBm－29dBm.
GSM850: 0dBm－33dBm
WCDMA 850： 0dBm—24dBm
EDGE 1900： 0dBm—26dBm
EDGE 850： 0dBm—27dBm

Maximum Power Limits

The peak output power of this mobile phone may not exceed 2 Watts as defined in Part 24E and 7 Watts as defined in Part 22H.

Antenna description

Technical parameters of the U1307 Mobile Phone antenna:

Item	Description
Frequency	1850MHz to1990MHz
	824 to 849 MHz

Input impedance	50 Ohm
VSWR	$\leqslant$3
Peak gain	$\geqslant$0 dBi
Rated power	40 W
Polarization	Linear

Applied voltages:

Normal Voltage: 3.7V

Low Voltage: 3.6V

High Voltage: 4.2V

Complete bill of material

Attachment

Complete Circuit Diagrams

Attachment

Instruction/Installation Manual

Attachment

Equipment label with FCC identification

Attachment

Means for Frequency Stabilization

The Voltage Controlled Temperature Compensated Crystal Oscillator (VCTCXO) provides the reference frequency for all RFIC synthesizers as well as clock generation functions within the MSM6245 IC. The oscillator frequency is controlled by the MSM6260 TRK_LO_ADJ pulse density modulated signal

Means for Limiting Modulation

In a GSM system, the input signal (voice for example) is sampled and coded in a vocoder, after channel coding, The digital signal is modulated onto the analog carrier frequency using Gaussian-filtered Minimum Shift Keying (GMSK). The modulation scheme is gaussian MSK (GMSK) with BT = 0,3. The modulation rate is 1 625/6 kbit/s (270,83 kbit/s). GMSK was selected over other modulation schemes as a compromise between spectral efficiency, complexity of the transmitter, and limited spurious emissions.

Description of Digital Modulation Techniques

   GSM is a digital system, so speech which is inherently analog, has to be digitized. The method employed by ISDN, and by current telephone systems for multiplexing voice lines over high speed trunks and optical fiber lines, is Pulse Coded Modulation (PCM). The output stream from PCM is 64 kbps, too high a rate to be feasible over a radio link. The 64 kbps signal,

although simple to implement, contains much redundancy. The GSM group studied several speech coding algorithms on the basis of subjective speech quality and complexity (which is related to cost, processing delay, and power consumption once implemented) before arriving at the choice of a Regular Pulse Excited -- Linear Predictive Coder (RPE--LPC) with a Long Term Predictor loop. Basically, information from previous samples, which does not change very quickly, is used to predict the current sample. The coefficients of the linear combination of the previous samples, plus an encoded form of the residual, the difference between the predicted and actual sample, represent the signal. Speech is divided into 20 millisecond samples, each of which is encoded as 260 bits, giving a total bit rate of 13 kbps. This is the so-called Full-Rate speech coding. Recently, an Enhanced Full-Rate (EFR) speech coding algorithm has been implemented by some North American GSM1900 operators. This is said to provide improved speech quality using the existing 13 kbps bit rate.

Because of natural and man-made electromagnetic interference, the encoded speech or data signal transmitted over the radio interface must be protected from errors. GSM uses convolutional encoding and block interleaving to achieve this protection. The exact algorithms used differ for speech and for different data rates. The method used for speech blocks will be described below.

Recall that the speech codec produces a 260 bit block for every 20 ms speech sample. From subjective testing, it was found that some bits of this block were more important for perceived speech quality than others. The bits are thus divided into three classes:

Class Ia 50 bits - most sensitive to bit errors

Class Ib 132 bits - moderately sensitive to bit errors

Class II 78 bits - least sensitive to bit errors

Class Ia bits have a 3 bit Cyclic Redundancy Code added for error detection. If an error is detected, the frame is judged too damaged to be comprehensible and it is discarded. It is replaced by a slightly attenuated version of the previous correctly received frame. These 53 bits, together with the 132 Class b bits and a 4 bit tail sequence (a total of 189 bits), are input into a 1/2 rate convolutional encoder of constraint length 4. Each input bit is encoded as two output bits, based on a combination of the previous 4 input bits. The convolutional encoder thus outputs 378 bits, to which are added the 78 remaining Class II bits, which are unprotected. Thus every 20 ms speech sample is encoded as 456 bits, giving a bit rate of 22.8 kbps.

To further protect against the burst errors common to the radio interface, each sample is interleaved. The 456 bits output by the convolutional encoder are divided into 8 blocks of 57 bits, and these blocks are transmitted in eight consecutive time-slot bursts. Since each time-slot burst can carry two 57 bit blocks, each burst carries traffic from two different speech samples.

Recall that each time-slot burst is transmitted at a gross bit rate of 270.833 kbps. This digital signal is modulated onto the analog carrier frequency using Gaussian-filtered Minimum Shift Keying (GMSK). GMSK was selected over other modulation schemes as a compromise between spectral efficiency, complexity of the transmitter, and limited spurious emissions. The

complexity of the transmitter is related to power consumption, which should be minimized for the mobile station. The spurious radio emissions, outside of the allotted bandwidth, must be strictly controlled so as to limit adjacent channel interference.

WCDMA or UMTS – as it is called throughout Europe – is a standard which has been developed to accommodate higher data rates to allow features like internet surfing, video telephony or video download. Even though WCDMA is focusing on high data rates it still supports simple features like a plain voice call or sending of SMS.

WCDMA is a CDMA system. CDMA stands for code division multiple access. This means that the available frequency channel is broken down by different code sequences that are multiplied by the user signals of the individual subscribers. All subscribers transmit on the same frequency and at the same time.

For WCDMA different base stations are distinguished by a different scrambling code, which makes cell planning a lot easier, since neighboring cells can re-use the same frequency! (However, the occupied "SNR" – or Signal to Noise Ratio is the limiting factor and characteristic for CDMA)

The data rate used by a terminal depends on spreading factor assigned to this particular terminal. If several terminals use the same spreading factor, the signals are distinguished through different code channels. At present the maximum data rate is 384 kbps. In the future it will be possible to combine several code channels to a multi-code link, allowing data rates up to 2 Mbps. However, when this is used the capacity of this frequency channel is used up, i.e. no other terminal can operate on this frequency channel. The reason for this is that there is no more "SNR" left for additional connections. This is the capacity issue indicated above.