ABSTRACT

Frequency Shift Keying or FSK is one of the most popular modulation schemes being used in digital radio. LMX2571 is a low power PLL with integrated VCO, it can support FSK modulation through programming or pins.

This application note will focus on how to apply data to implement analog FM and pulse-shaped FSK modulation with LMX2571 in FSK I2S mode.

1 Introduction

Frequency Shift Keying or FSK is one of the most popular modulation schemes being used in digital radio. This is because this modulation is very easy to implement in hardware. For a simple two-level FSK, the serial data stream is used to modulated the VCO directly. When the data bit is 0, output frequency is $f_0$. Output frequency becomes $f_1$ when the data bit is 1. The difference between $f_0$ and $f_1$ is the so-called frequency deviation. The drawback of this FSK is the occupied bandwidth will be very large. In order to
increase spectrum efficiency, the data bit is usually pulse-shaped before being used to modulate the VCO. In this case, the discrete-level digital data stream will pass through a digital filter, usually a Raised Cosine filter or a Gaussian filter. The output of the filter becomes a continuous-level analog signal, no more abrupt jump from 0 to 1 and vice versa. Instead of modulating the VCO, another way to create FSK modulation is to "modulate" the N-counter of the PLL. This method is called Direct Digital FSK Modulation. In this case, the frequency deviation is obtained by changing the N-counter value.

LMX2571 is a low power PLL with integrated VCO, it can support direct digital FSK modulation through programming or pins. In FSK SPI and FSK PIN mode, the LMX2571 supports discrete level FSK modulation that allows the N-counter value to be modulated between 2-, 4- or 8-different values.

LMX2571 can also support arbitrary level FSK or pulse-shaped FSK in FSK SPI FAST and FSK I2S mode. LMX2571 does not have a machine to manipulate pulse-shaping to incoming data bits. Pulse-shaping of the data bits has to be done externally with a digital processor before applying it to LMX2571.

2 Theory of Operation

The operation of direct digital FSK modulation is similar to the operation of a data converter system. First of all, the FSK/FM modulation signal gets sampled. The sampled amplitudes are transformed to a stream of digital codes. This analog-to-digital conversion is ideally handled with a digital processor.

When the high frequency carrier gets frequency modulated, we are indeed interested in frequency deviation vs time instead of voltage vs time. So before A-to-D conversion, we should "scale" the voltage of the modulation signal to frequency deviation. For example, assume the modulation signal is a sine wave with amplitude swing equals 1 V, and 1 V is equivalent to 2 kHz frequency deviation. On every sampling point, the equivalent frequency deviation is transformed to a 16-bit word.

LMX2571 acts as the digital-to-analog converter in this system. It accepts the stream of 16-bit word and then reconstructs the desired frequency deviation.
3 Analog FM Modulation Example

In this example, LMX2571 is used to generate a FM modulated carrier (similar to an analog FM radio transmitter). The "modulation signal" is a pure sine wave tone.

3.1 Design Requirements

OSCin frequency = 80 MHz
RFout frequency = 500 MHz
FM modulation frequency = 1 kHz
FM frequency deviation = 2 kHz
Over-sampling factor = 192x
Over-sampling rate = 192 kHz
Fractional denominator, DEN = $2^{24}$

3.2 Design Procedure

3.2.1 Step 1

Calculate all the frequencies in each functional block of LMX2571.

3.2.2 Step 2

For a complete modulation signal period, there are 192 sample points. In other words, on every $360° / 192 = 1.875°$, the modulation signal gets sampled once. The frequency deviation of each sample point is equal to $2 kHz \times \sin (phase \times \frac{\pi}{180°})$, where phase = $0°, 1.875°, 3.75°, 5.625°, \ldots, 358.125°$. 

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Analog FM Modulation Example

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Frequency Shift Keying with LMX2571
3.2.3 Step 3

Use Equation 4 and 6 in the LMX2571 datasheet, calculate the corresponding code for each sample point.

<table>
<thead>
<tr>
<th>TIME</th>
<th>FREQUENCY DEVIATION</th>
<th>CORRESPONDING FSK STEPS</th>
<th>BINARY EQUIVALENT</th>
</tr>
</thead>
<tbody>
<tr>
<td>t₁</td>
<td>0 Hz</td>
<td>0</td>
<td>0000 0000 0000 0000</td>
</tr>
<tr>
<td>t₂</td>
<td>65.438 Hz</td>
<td>69</td>
<td>0000 0000 0100 0101</td>
</tr>
<tr>
<td>t₃</td>
<td>130.806 Hz</td>
<td>137</td>
<td>0000 0000 1000 1001</td>
</tr>
<tr>
<td>t₄</td>
<td>196.034 Hz</td>
<td>206</td>
<td>0000 0000 1100 1110</td>
</tr>
<tr>
<td>...</td>
<td>...</td>
<td>...</td>
<td>...</td>
</tr>
<tr>
<td>t₁₄₅</td>
<td>−2000 Hz</td>
<td>63438</td>
<td>1111 0111 1100 1110</td>
</tr>
<tr>
<td>...</td>
<td>...</td>
<td>...</td>
<td>...</td>
</tr>
<tr>
<td>t₁₉₂</td>
<td>−65.438 Hz</td>
<td>65466</td>
<td>1111 1111 1011 1010</td>
</tr>
</tbody>
</table>

3.2.4 Step 4

At last, feed the binary code to LMX2571. See Section 7.3.13 in the LMX2571 datasheet for the details on FSK I2S mode.

Figure 4. FSK I2S Interface

3.3 Application Curves

![Figure 5. Analog FM Spectrum View](image)

![Figure 6. Analog FM Modulation-Domain View](image)

Figure 5 shows the transmit spectrum while Figure 6 shows the carrier is moving around at a rate of 1 kHz and with a peak frequency deviation of 2 kHz.
4 Raised Cosine 4FSK Example

This is a pulse-shaped FSK example. The "modulation signal" is a 4-level raised cosine filtered FSK signal.

4.1 Design Requirements

OSCin frequency = 80 MHz
RFout frequency = 500 MHz
4FSK modulation baud rate = 4.8 kSps
Raised Cosine pulse-shaping filter, BT = 0.2
FSK frequency deviation = 648 Hz and 1944 Hz
Over-sampling factor = 20x
Over-sampling rate = 96 kHz
Fractional denominator, DEN = $2^{24}$

4.2 Design Procedure

Basically the design procedures are same as the previous example, we have to go through Step 1 to Step 4. The difference in this example is the modulation signal is not a simple sine wave but a stream ofRaised Cosine-shaped 4FSK signal. This signal is first over-sampling at a rate of 96 kHz. Again, use Equation 4 and 6 to convert the corresponding frequency deviation at each sample point to I2S code.

Figure 7. Raised Cosine 4FSK Signal
4.3 Application Curves

Because the FSK signal has been pulse-shaped, the occupied bandwidth is very minimal, as shown in Figure 8. The advantage of using direct digital modulation is revealed in Figure 9 where we can see that the modulation quality is excellent. Furthermore, this technique is immune to Process, Voltage rail and Temperature variation because it is a closed-loop digital technique.

5 Gaussian 4FSK Example

This is another pulse-shaped FSK example. The "modulation signal" is a 4-level Gaussian filtered FSK signal. In this example, the frequency deviation is larger. Fractional denominator, DEN, is deliberately made smaller in order to achieve a larger frequency deviation.

5.1 Design Requirements

- OSCin frequency = 80 MHz
- RFout frequency = 470 MHz
- 4FSK modulation baud rate = 125 kSps
- Gaussian pulse-shaping filter, BT = 0.4
- FSK frequency deviation = 17 kHz and 51 kHz
- Over-sampling factor = 10x
- Over-sampling rate = 1.25 MHz
- Fractional denominator, DEN = 8000000

5.2 Design Procedure

Same as Section 4.2 except for the modulation signal is filtered with a Gaussian filter.
5.3 Application Curves

Even with a higher data rate as well as a higher frequency deviation, the modulation spectrum can meet radio standard's spectrum mask requirement.
6 Direct Digital FSK Modulation Design Considerations

6.1 Loop Bandwidth Selection

As a general rule of thumb, the loop bandwidth has to be greater than or at least compatible with the sampling rate so that it does not degrade the data. Remember that the modulation is implemented by changing the N-counter values, i.e., we are modulating the PLL instead of VCO. Because loop filter behaves as a low pass filter to the PLL, out-of-band modulation will get distorted. As a consequence, loop bandwidth should be greater than the sampling rate. If the modulation signal is a square wave, as in the case of discrete FSK modulation, loop bandwidth should be at least twice the baud rate.

6.2 Fractional Denominator Selection

From the datasheet of LMX2571, it is required that the fractional denominator has to be equal to $2^{24}$ whenever FSK is enabled. This constrain will therefore limit the maximum achievable frequency deviation. For example, in Section 4, the maximum achievable frequency deviation is just 31.249 kHz. It is indeed possible to make the frequency deviation larger with a smaller fractional denominator value.

The way that FSK works in LMX2571 is by adding or subtracting the necessary FSK steps from the fractional numerator of the carrier. For example, if the numerator of a particular carrier frequency is 1000 and the necessary FSK step value for a 1 kHz deviation is 1001. To get +1 kHz deviation, the instantaneous numerator value is 2001. This is a valid numerator value so we will be able to get +1 kHz deviation from the carrier. However, we are not able to get –1 kHz deviation because the instantaneous numerator value is an invalid number of –1. This situation is already taken care with when denominator is equal to $2^{24}$.

In order to get a larger frequency deviation, we can make the denominator smaller, as long as the numerator of the carrier is greater than the necessary FSK step value. For instance, if we put denominator equals to 8000000 instead of $2^{24}$, the maximum frequency deviation we can get becomes 65.534 kHz. A practical example is already illustrated in Section 5.

At last, there are a couple of things to remember.

- Although the fractional denominator is 24-bit, we only use 16-bit to represent the frequency deviation (or FSK step).
- The maximum FSK step value for positive frequency deviation is equal to $2^{15} - 1$.
- FSK step values are expressed in 2’s complement so both negative and positive deviation can be made.

6.3 Over-sampling Rate Selection

One of the advantages of direct digital FSK modulation is the frequency deviation is linear over the entire frequency band of interest. That is, we can get the same frequency deviation at 100 MHz or 1000 MHz. This is not possible if we were to modulate the VCO because the VCO gain (aka $K_{VCO}$) varies with VCO frequencies. However, the adoption of direct digital FSK modulation requires a higher signal processing power capability from the system. This is because we need over-sampling at a rate much higher than the baud rate of the modulation signal.

When the N-counter gets modulated at a rate of M kHz, there are sidebands appear at multiple of M kHz offset from the carrier. The magnitude of the closest sideband gets 12 dB down whenever the sampling rate is doubled. As such, around 100 kHz is the minimum sampling rate we recommend. As these sidebands can be reduced by the loop filter, if a higher modulation distortion or FSK error is acceptable, we can make the loop bandwidth smaller than the sampling rate in order to reduce the amplitude of these sidebands.
7 Conclusion
This application note demonstrates how to use the FSK feature of LMX2571 to implement analog FM modulation and pulse-shaped FSK modulation. Direct digital FSK modulation design considerations are also addressed.