**CML Microcircuits** *COMMUNICATION SEMICONDUCTORS*

D/649/6 February 2013

# **Features**

- **Multiple Codec Modes, 16 to 128 kbps** 
	- - **Full duplex ADM and CVSD**
	- - **Full duplex PCM: µ-law, A-law, Linear**
	- - **Configurable ADM time constants**
	- - **Dual channel transcoder/decoder mode**
	- **High Performance Digital Architecture**
- **Low Power: 2.4mA at 3.0V typ.**
- **2.7V 5.5V Supply**
- **Data Clock Recovery**
- **Programmable Voice Activity Detector (VAD)** 
	- - **Adjust threshold level and attack/decay time**
	- - **Use to powersave on low signal level**
	- - **Silence/blank low level signals**
	- **Programmable Digital Scrambler**
- **Flexible Interfaces** 
	- - **8-bit and 16-bit burst data with sync strobe**
	- - **1 bit serial data with clock**
	- - **Host serial control/data interface**
- **Internal and External Sample Clocking**
- **Programmable Filters** 
	- - **Encoder mic input ADC anti-alias**

**CMX649**

**ADM Codec**

- - **Decoder audio out DAC anti-imaging**
- **Low Noise Differential Mic Input Amp**
- **Programmable Analog Interface Gain** 
	- - **Microphone in**
	- - **Decoder audio out**
	- - **Sidetone path**

# **Applications**

- **Low Cost Digital Cordless Headset**
- **Personal Area Network (PAN) Voice Link**
- **Digital Cordless Telephone**
- **Wireless Digital PBX**
- **Full Duplex Digital Radio Systems**
- **Time Division Duplex (TDD) Systems**
- **Portable Digital Voice Communicator**
- **Digital Voice Delay**



# **1. Brief Description**

The CMX649 Adaptive Delta Modulation (ADM) Voice Codec provides full duplex ADM, companded (µ/Alaw) PCM and linear PCM codec and transcoder functions for cost effecive, low power, wireless voice applications. Selectable modes and algorithms support many requirements. Robust ADM coding (e.g. CVSD) reduces host protocol and software burdens, eliminating forward error correction, framing protocols and algorithm processing. Dual transcode/decode mode supports multichannel applications.

Integrated filter responses adjust independently of 16kbps to 128kbps codec data rates. Codec sample clocks are externally applied or internally generated. High performance analog interfaces and sidetone include digital gain controls. Encoder and decoder voice activity detectors support powersaving.

The CMX649 ADM Voice Codec supports 2.7V to 5.5V operation and is available in 20-pin SOIC (D3) and TSSOP packages (E3) packages.

# **CONTENTS**







**Figure 1 Block Diagram** 

# **3. Signal List**



**Notes:** I/O = Input/Output NC = No Connection

© 2013 CML Microsystems Plc 649/6

# **4. External Components**



#### **Figure 2 Recommended External Components**



#### **Notes:**

- 1. C1 C4 and R1 R4 set the microphone amplifier gain and frequency response. The values shown set the gain to unity and the low and high -3dB frequency rolloff points to approximately 150Hz and 15kHz respectively.
- 2. DC blocking capacitor for driving a speaker from an external speaker amplifier. The value shown is based on a  $32\Omega$  impedance speaker where the highpass rolloff frequency is set to approximately 150Hz.
- 3. V<sub>DD</sub> decoupling capacitor.
- 4. Bias decoupling capacitor.
- 5. A 8.192MHz Xtal/Clock input will yield exactly 16kbps/32kbps/64kbps internally generated data clock rates.
- 6. To achieve good noise performance,  $V_{DD}$  and  $V_{BIAS}$  decoupling and protection of the signal path from extraneous in-band signals are very important. It is recommended that the printed circuit board is laid out with a ground plane in the CMX649 area to provide a low impedance connection between the  $V_{SS}$  pin and the  $V_{DD}$  and  $V_{BIAS}$  decoupling capacitors.

# **5. General Description**

The CMX649 encodes and decodes analog audio signals to/from ADM, Linear PCM, µ-law PCM or A-law PCM. It has programmable clock dividers that enable it to use an 8.000 MHz to 16.384 MHz crystal (or a 2.048 MHz to 16.384 MHz external clock source) and to sample the data over a large range of data rates. Programmable current sources for on-chip op-amps enable the overall power consumption to be optimised for any given supply voltage and clocking scheme, thus achieving extremely low working power levels.

Anti-Alias Image filters and gain controls are fully programmable. All the time constants and other parameters of the ADM can be programmed for optimum performance.

The CMX649 also includes a Microphone Amplifier, Data Clock Recovery, Data Scrambler/De-Scrambler and Voice Activity Detector (VAD) circuits.

All of these parameters are controlled via C-BUS.

#### **5.1 Block Descriptions**

The CMX649 contains a full duplex speech codec supporting common Adaptive Delta Modulation (ADM) and non-linear PCM coding algorithms. In addition, it supports linear PCM coding for DSP interface applications. This codec offers simple interface and application, yet is configurable to support a wide variety of speech quantisation systems.

# **5.1.1 ADM Coding Engine**

ADM is a differential waveform coding technique predominantly applied to speech. Figure 3 illustrates the ADM encoder employed. The device is for speech quantising applications and is based on popular Continuously Variable Slope Delta (CVSD) encoder approaches, with optional modifications and improvements configurable through the ENCODE and DECODE ADM CONTROL Registers (\$E1 and \$D1). Optional second order integration in the feedback loop provides improved speech quality at a given bit rate or similar quality at a lower bit rate. Toll quality is achieved at bit rates much lower than for PCM. The decoder is embedded in the encoder, as is the case with most differential encoders. Note the symmetry between the encoder and decoder of Figures 3 and 4 respectively. The signal flows for ADM are shown in bold.



**Figure 3 ADM Encoding** 



# **Figure 4 ADM Decoding**

The estimator integrators (principal and second) as well as the step size decay (companding integrator) have programmable time constants*.* Additionally, the minimum and maximum step height and the depth of the delay register are programmable via preset values in the DECODE and ENCODE ADM CONTROL Registers (\$D1 and \$E1) to support a wide variety of different ADM algorithms including CVSD of Bluetooth™ version 1.1. The switches in Figures 3 and 4 are controlled by the ENCODER and DECODER MODE and SETUP Registers (\$E0 and \$D0). Various signal flows are possible to allow standard ADM and PCM encoding and decoding as well as transcoding either direction between ADM and PCM (e.g. Figures 7 and 8). Additionally, several summing options are possible. In the decoder a PCM and ADM input stream may be summed – note that this requires at least one of the streams to be input via C-BUS. In the encoder a PCM input stream may be summed with the ADM estimate causing the encoded ADM bit stream to represent the sum of the analog input and linear PCM stream input over C-BUS.





**Figure 5 PCM Encoding** 

The output of the first or principal estimator integrator in Figures 3 and 4 is linear PCM. By decimating and filtering this signal it is possible to obtain a linear PCM representation, as shown in Figures 5 and 6. Employing either 8:1 or 4:1 decimation filters provides about 30dB attenuation of out of band quantisation noise prior to decimation. The ADM coding engine, which suppresses out of band noise by roughly 20dB, provides (in conjunction with the decimating filter) an overall out of band suppression of approximately 50dB. Using second order ADM at 64kbps with the 8:1 decimation filter provides better than toll quality linear speech samples. Accordingly, 8k samples/sec linear PCM encoder performance can be enhanced when the ADM codec second order integrator is enabled and the ADM codec is operated at the maximum rate. Decoding PCM simply requires interpolation and filtering to compensate for sin(x)/x roll-off of zero holding the PCM samples. The interpolation ratio can be programmed to 4 or 8.

2013 CML Microsystems Plc 8 D/649/6



**Figure 6 PCM Decoding** 







(note that the decoder also can be configured to do this function and in this example all data is read and written via C-BUS registers \$EA (\$DA) and \$E7 (\$D7) respectively (for decoder)).





(note that the encoder can also be configured to do this function and in this example all data is read and written via C-BUS registers \$D6 (\$E6) and \$D8 (\$E8) respectively (for encoder)).

2013 CML Microsystems Plc 9 D/649/6

#### **5.1.4 Non-Linear Instantaneous Companding**

When using the device over its standard PCM codec style interface, instantaneous companding can be enabled to cut in half the PCM word size. Either µ-law or A-law type companding algorithms are provided and use 16-chord piecewise linear approximations. Essentially the companded 8-bit PCM word is a simple floating-point representation with a sign bit, a 3-bit exponent and a 4-bit mantissa. This approach yields toll quality speech at reduced data rates.

#### **5.1.5 Digitally Controlled Amplifiers**

There are three Digitally Controlled Amplifiers (DCA) on-chip, which are used to set the signal levels for transmit-audio-in, side-tone-audio, and receive-audio-out (volume control). The transmit-audio DCA is adjustable in 0.5dB steps over a  $+7.5$ dB to  $-7.5$ dB range. The side-tone DCA is adjustable in 6.0dB steps over a 0dB to -21.0dB range. Side-tone audio is added to the audio output signal via an operational amplifier configured as a summing amplifier. This feeds the receive-audio DCA, which is adjustable in 1.5dB steps over a +12.0dB to –33.0dB range.

#### **5.1.6 Microphone Amplifier**

The input amplifier is a high gain low-noise operational amplifier capable of interfacing with a variety of different microphones. Figure 9 is a simplified schematic showing the external components required for typical application with an electret condenser microphone. Typical values for R1, R3, C1 and C3 should be set according to microphone sensitivity requirements, those shown are for unity gain. Note also that the microphone biasing resistors (R5 and R6) are microphone specification dependent.



#### **Figure 9 Electret Microphone - Input Amplifier Schematic**



#### **5.1.7 Programmable Anti-alias/image SC Filters**

The anti-aliasing (AAF) and anti-imaging (AIF) switched capacitor (SC) filters have a programmable cut-off frequency to accommodate different input signal bandwidths. Typically, the audio filter bandwidth should be programmed to be 1/10<sup>th</sup> of the ADM bit rate (or lower) for "toll" (or better) quality audio reconstruction. For "communications" quality, the audio bandwidth may approach 1/6<sup>th</sup> of the ADM bit rate for ADM rates below 20kbps. The anti-alias/image SC filter bandwidth is programmed directly via C-BUS commands to the AAF/AIF BANDWIDTH Register (\$61). Additionally, the switched capacitor clock frequency can be altered via C-BUS commands to the CLK DIVIDER CONTROL Register (\$72). Typically, the CLK DIVIDER CONTROL Register should be programmed to provide a 256kHz SC filter clock. Altering the SC filter clock from the recommended 256kHz frequency proportionally scales the frequency axis in the plot below:



**Figure 10a Typical Anti-Alias/Image Filter Frequency Response** 



**Figure 10b Typical Anti-Image Filter Frequency Response** 

# **5.1.8 Data Clock Recovery**

Data from the RX DATA pin is driven into a comparator to remove amplitude variations. The output of the comparator is a logic signal that can be inverted by setting the appropriate control bit in the SCRAMBLER CONTROL Register (\$71). Using the output of the comparator, the clock recovery block can be enabled to generate a phase-locked clock equal to the CVSD data rate, which is used to clock data from the RX DATA pin into the decoder. The recovered clock frequency is controlled by the CLK DIVIDER CONTROL Register (\$72). If the clock recovery block is bypassed, data must then be applied which is synchronised to the clock on the RX CLK pin (either internally generated or externally applied).

External ADM rate bit clocks can be used for both the encoder and decoder paths and do not require use of the clock recovery PLL. Externally applied clocks act directly as the ADM sample clocks and should be generated with little jitter for best performance. Please note that the maximum usable frequency of externally applied bit clocks is 1/60<sup>th</sup> of the frequency of the output of the internal bit clock prescaler.

The clock recovery circuit is normally applied to the decoder. However, it is possible to use the recovered clock for the encoder section as well. This supports systems where the base unit is using an internal clock or local external clock for transmit and clock recovery for the decoder clock. The remote unit can then be configured to use the recovered clock for both encode and decode. Internal data clocks for the encoder and decoder can also be selected for data input and output control.

#### **5.1.9 Data Scrambler/De-scrambler**

The scrambler receives digital data from the encoder. It is implemented with a 10-bit programmable linear feedback shift register (LFSR) allowing a choice of various maximal length scrambling codes. The scrambler, also known as a randomizer, provides not only a level of communication security, but may also help reduce the occurrence of abnormally long strings of 1s or 0s.

The de-scrambler receives the scrambled data from the data slicer and de-scrambles it to the original data as long as the selected LFSR maximal length sequence is the same as that in the transmitting scrambler. The de-scrambler block has the same configuration as the scrambler and is self-synchronizing. Both the scrambler and de-scrambler can be bypassed.

Nine example maximal length codes are represented below through their polynomial coefficients which can be directly programmed in Bits 9-0 of the SCRAMBLER CONTROL Register (\$71):



#### **5.1.10 Voice Activity Detector (VAD)**

The VAD function is implemented with an energy detector circuit. This circuit consists of an absolute value function, an integrator and a threshold detector. The threshold detector level and the integrator time constants (i.e. attack and decay time control) are user programmable via the DECODE and ENCODE VAD THRESHOLD Registers (\$D2 and \$E2) and the DECODER and ENCODER MODE AND SETUP Registers (\$D0 and \$E0). Referring to Figure 11, the input to the VAD comes from the PCM signal. The signal is rectified and averaged with a lossy integrator. The output of the integrator is compared to the VAD threshold to derive the logic signal VAD\_OUT. If VAD\_OUT is a logic one, signal energy greater than the threshold is present. If VAD\_OUT is a logic zero, signal energy is below the threshold. When the VAD "trip" threshold has been reached, the amplitude required to clear this VAD state is set internally to half of the "trip" threshold. For example, if the VAD "trip" threshold is set to 100mV then the "clear" threshold will be 50mV. This hysteresis is provided to minimise chattering and is not user-adjustable. Attack and decay times for the decoder VAD and encoder VAD can be independently controlled via the DECODER and ENCODER MODE AND SETUP Registers (\$D0 and \$E0). Typical attack and decay times used for detecting voice activity are 5ms and 150ms, respectively. The energy levels may be read from DECODE and ENCODE VAD LEVEL OUTPUT Registers (\$D4 and \$E4) for the decoder and encoder and used to adaptively set the detector threshold value by observing the energy level of background noise.



**Figure 11 VAD Block Diagram** 

© 2013 CML Microsystems Plc 13 D/649/6

# **Address/Commands**

Instructions and data are transferred, via C-BUS, in accordance with the timing information given in Figure 12.

 Instruction and data transactions to and from the CMX649 consist of an Address/Command (A/C) byte followed by either:

- (i) a further instruction or data (1 or 2 bytes) or  $(ii)$  a status or Rx data reply (1 or 2 bytes)
- a status or Rx data reply (1 or 2 bytes)



# **Write Only C-BUS Registers**



<sup>© 2013</sup> CML Microsystems Plc 15 15 2013 CML Microsystems Plc



# **Read Only C-BUS Registers**



<sup>© 2013</sup> CML Microsystems Plc 16 16 2013 CML Microsystems Plc



#### **5.2.1 Write Only Register Description**

#### **5.2.1.1 GENERAL RESET (\$01)**

The reset command has no data attached to it. Application of the GENERAL RESET sets all write only register bits to 0.

#### **5.2.1.2 AAF/AIF BANDWIDTH Register (\$61)**

**AAF Bypass**  When this bit is set to logic 1 the anti-alias filter is bypassed.

**(Bit 7)** 

**AAF Bandwidth (Bits 6 – 4)**  The  $-3$ dB cutoff frequency of the anti-alias filter is controlled by bits  $4-6$ . The filter shape is not altered other than to move the cutoff frequency.



**AIF Bypass**  When this bit is set to a logic 1 the anti-image filter is bypassed.

**(Bit 3)** 

**AIF Bandwidth (Bits 2 – 0)**  The –3dB cutoff frequency of the anti-image filter is controlled by bits 0-2. The filter shape is not altered other than to move the cutoff frequency.



# **5.2.1.3 VOLUME/SIDETONE LEVEL Register (\$62)**

**Volume Level (Bits 7 – 3)**  The five most significant bits in this register are used to set the gain of the volume control according to the table below:



**Sidetone Level (Bits 2 – 1)** 

These bits control the gain of the sidetone signal coming from the AAF output to be summed in with the decode signal at the input to the AIF.



**Sidetone Enable (Bit 0)**  When this bit is a logic 1 the sidetone path is enabled with the gain setting controlled as shown above. When this bit is logic 0 the sidetone path is disabled.

# **5.2.1.4 AUDIO INPUT LEVEL CONTROL Register (\$63)**

**Audio Input Level Control (Bits 7 – 3)**  These bits are used to set the gain of the Digitally Controlled Amplifier (DCA) at the output of the microphone amplifier.



**Reserved (Bits 2– 0)**  These bits are reserved and should be set to a logic 0.

# **5.2.1.5 POWER CONTROL 1 Register (\$64)**

**AAF Power Control (Bit 7)**  This bit is reserved and should be set to a logic 0

**AAF Power Control**  This bit is dedicated to power/current control for the AAF. Note: It is necessary to keep the power level set to one of the "ON" settings when the AAF is bypassed.

**(Bit 6)** 



2013 CML Microsystems Plc 19 D/649/6



**AIF Power Control (Bit 4)**  This bit is dedicated to power/current control for the AIF and the Sidetone DCA. Note: It is necessary to keep the power level set to one of the "ON" settings when the AIF is bypassed.



**Encode DAC Power Control**  These bits are dedicated to power/current control for the Encode DAC.

**(Bits 3 – 2)** 



**Decode DAC Power Control**  These bits are dedicated to power/current control for the Decode DAC.

**(Bits 1 – 0)** 



# **5.2.1.6 POWER CONTROL 2 Register (\$65)**

**MIC AMP Power Control**  These bits are dedicated to power/current control for the Microphone Amplifier.

**(Bits 7 – 6)** 



**AUDIO DCA Power Control** 

These bits are dedicated to power/current control for the Audio Input Digitally Controlled Amplfier.

**(Bits 5 – 4)** 



**VOLUME DCA Power Control (Bits 3 – 2)**  These bits are dedicated to power/current control for the Volume Digitally Controlled Amplifier.



**XTAL Power Save (Bit 1)**  When this bit is a logic 1 the one-pin crystal oscillator circuit is powered down.

**ANALOG Enable (Bit 0)**  When this bit is set to a logic 1 all of the analog circuitry (register \$64 and bits 7-2 of register \$65) is enabled. When this bit is set to a logic  $\overline{0}$  all of the analog circuitry is powered down (on-chip bandgap reference is powered down). This is equivalent to setting all of the bits of register \$64 and bits 7-2 of register \$65 to a logic 0. So to enable the AAF pabk, register \$64 bit 6 and register \$65 bit 0 must both be set to logic 1. Note that these bits control power to their respective blocks and that a signal path may still exist even if the block is powersaved.

#### **5.2.1.7 CODEC MODE CONTROL Register (\$70)**

**Reserved**  These bits are reserved and should be set to a logic 0.

## **(Bits 7 – 3)**

# **CODEC MODE**

**(Bits 2 – 0)** 



#### **5.2.1.8 SCRAMBLER CONTROL Register (\$71)**

Scrambler Enable (Bit 15)	Setting this bit to a logic 1 enables the scrambler.				
Reserved (Bit 14)	Reserved for future use. Set to '0'.				
Scrambler <b>Output Invert</b> (Bit 13)	Setting this bit to a logic 1 inverts the scrambler output polarity.				
<b>De-Scrambler</b> Enable (Bit 12)	Setting this bit to a logic 1 enables the de-scrambler.				

 <sup>2013</sup> CML Microsystems Plc 21 D/649/6



# **5.2.1.9 CLK DIVIDER CONTROL Register (\$72)**



**(Bits 12 – 11)** 



These bits control the internal switched capacitor filter clock divider.

**Filter Clock Divider (Bits 10 – 8)** 



#### 2013 CML Microsystems Plc 22 D/649/6

#### **Bit Clock Pre-**These bits control the bit clock pre-scaler.

#### **Scaler (Bits 7 – 6)**



**Decode Bit Clock Divider (Bits 5 – 3)** 

These bits control the decode bit clock divider.



**Encode Bit Clock Divider (Bits 2 – 0)** 

These bits control the encode bit clock divider.

**Bit 2 Bit 1 Bit 0 Divider Ratio**   $\begin{array}{|c|c|c|c|c|}\n\hline\n0 & 0 & 1 & 2.000 \\
\hline\n0 & 0 & 1 & 2.000\n\end{array}$  $\begin{array}{|c|c|c|c|}\n\hline\n0 & 1 & 2.000 \\
\hline\n1 & 0 & 2.250\n\end{array}$ 0 1 0 2.250 0 1 1 2.625 1 0 0 3.000 1 0 1 3.125  $1 \mid 1 \mid 0 \mid 3.375$ 1 | 1 | 1 | 3.500

The CLK DIVIDER CONTROL Register should be programmed to provide a 256kHz nominal SC filter clock. Programming the SC filter clock to different frequencies will introduce a proportionate frequency shift in the configured anti-alias/image SC filter frequency responses. See also section 6.3.

The audio filter clock divider should be programmed to set the audio filter clock as near as possible to 256kHz, via selection of the XTAL frequency and the Filter Prescaler and Filter Divider settings.

The encoder and decoder ADM bit rate clocks should be programmed to the desired ADM bit rate or PCM sample rate, multiplied by the interpolation/decimation setting of the PCM filter. The PCM filter can be programmed to run at either 4x or 8x the PCM sample rate depending on the corresponding setting in the encode/decode processors.

The encoder and decoder ADM bit rate clocks are further divided by a constant factor of 64 (unless the PLL is enabled in which case the average is near 64 but can pull off slightly depending on the reference source).

2013 CML Microsystems Plc 23 D/649/6

# **5.2.1.10 CLK SOURCE CONTROL Register (\$73)**



Note that a system clock or crystal is always required on the XTAL/CLK pin, in order to generate the various internal timing signals, even when Rx and Tx Clocks are recovered from the RX DATA pin.

Internally Generated from decode clock.

**Data Filter Bypass (Bit 3)**  Setting this bit to a logic 1 bypasses the data filter and inputs the RX DATA signal directly into the data slicer.

**Data Filter and Slicer Power Control (Bits 2 – 1)** 



When the Data Filter and Slicer are powered off, the RX DATA input pin signal must conform to logic level amplitudes. When operating the device in buffered I/O modes, the Data Filter and Slicer should be powered off.

**Data Filter Bandwidth (Bit 0)**  Setting this bit to a logic 1 forces the data filter to narrow bandwidth mode.

# **5.2.1.11 CODEC INTERRUPT CONTROL Register (\$81)**

**Encoder Control (Bits 7 – 4)** 



#### **Decoder Control (Bits 3 – 0)**



These bits are dedicated to power/current control for the data filter and slicer.

# **5.2.1.12 DECODER MODE AND SETUP Register (\$D0)**

**Decimation Rate (by 4/8) (Bit 15)**  The decoder PCM filter functions as an interpolator for the DAC when PCM words are being received by the decoder and as a decimator when the decoder is receiving delta modulation. In the case where delta modulation is received, transcoded PCM values are available in the DECODE LINEAR PCM OUTPUT Register (\$D6) at the decimation rate. When PCM is received the device can be set to transcode to an ADM stream available in the DECODE ADM OUTPUT Register (\$DA) at the interpolated rate. A logic 1 sets the interpolation (decimation) rate to  $4$  (1/4<sup>th</sup> the bit rate). A logic 0 sets the interpolation (decimation) rate to 8 ( $1/8<sup>th</sup>$  the bit rate).

![](_page_25_Picture_220.jpeg)

Allows selection of the input to the PCM rate converting filter.

![](_page_25_Picture_221.jpeg)

If PCM filter interpolates, the decoder can digitally transcode a PCM signal to ADM. If PCM filter decimates, the decoder can digitally transcode an ADM signal to PCM.

**ADM Input Select (Bits 12 – 11)** 

![](_page_25_Picture_222.jpeg)

![](_page_25_Picture_223.jpeg)

![](_page_26_Picture_217.jpeg)

is reserved and should be set to a logic 0.

**Decoder VAD Decay Time Constant** 

**(Bits 7 – 5)** 

Allows selection of the Voice Activity Detector decay time constant.

![](_page_26_Picture_218.jpeg)

**Decoder VAD Attack Time Constant (Bits 4 – 3)** 

Allows selection of the Voice Activity Detector attack time constant.

![](_page_26_Picture_219.jpeg)

**Decoder VAD** 

Allows selection of the Voice Activity Detector output source.

**Output Source** 

![](_page_26_Picture_220.jpeg)

**(Bits 2 – 1)** 

![](_page_26_Picture_221.jpeg)

**Reserved**  This bit is reserved and should be set to a logic 0.

**(Bit 0)** 

**5.2.1.13 DECODE ADM CONTROL Register (\$D1)** 

**Syllabic Time**  Step size integrator Loss Coefficient: allows selection of syllabic time constant.

![](_page_26_Picture_222.jpeg)

![](_page_27_Picture_228.jpeg)

![](_page_27_Picture_229.jpeg)

![](_page_27_Picture_230.jpeg)

**Companding Rule (Bits 9 – 8)** 

This is the number of consecutive ones or zeros that must occur for the step size to be adjusted.

![](_page_27_Picture_231.jpeg)

**Estimator Integrator Time Constant (Bits 7 – 5)** 

Allows selection of the estimator integrator time constant.

![](_page_27_Picture_232.jpeg)

**Second Order Estimator Time Constant (Bits 4 – 3)** 

Allows selection of the second order estimator time constant.

![](_page_27_Picture_233.jpeg)

**Zero Selection (Bits 2 – 1)** 

When second order integration is used, a zero can be inserted to help encoder stability. Not generally used in the decoder unless set to digitally transcode from PCM to ADM.

![](_page_27_Picture_234.jpeg)

**Input** 

![](_page_28_Picture_198.jpeg)

#### **5.2.1.14 DECODE VAD THRESHOLD Register (\$D2)**

![](_page_28_Picture_199.jpeg)

(DAC Full Scale Reference Voltage) Register Value =  $\frac{\text{(Signal Detection Threshold)} \cdot 2}{\text{Total REALP}}$  $=\frac{Signal Detection Threshold) \cdot 2^{15}}{Simplifying this.}$ 

# **5.2.1.15 DECODE OFFSET LEVEL Register (\$D3)**

**Decode Offset**  For normal Decoder operation this register should be set to logic 0.

**(Bits 15 – 0)**  These bits allow for an offset amount to be directly programmed. This offset amount is useful in trimming out offsets that may occur in the on-chip analog circuitry. The number format is 2's complement and ranges from \$8000 through \$0000 to \$7FFF (-32768 to 32767).

The equation for the Offset value is:

(DAC Full Scale Reference Voltage) Register Value =  $\frac{(Offset Voltage) \cdot 2}{P \cdot 2P \cdot R \cdot 2P \cdot R \cdot R \cdot 2P \cdot R}$  $=\frac{\text{(Offset Voltage)} \cdot 2^{18}}{2 \times 10^{-18} \text{ m} \cdot 10^{-18} \text{ m}^2}$ 

The programmed offset will be summed with the decoder output signal.

# **5.2.1.16 DECODE LINEAR PCM INPUT Register (\$D7)**

**Decode Linear PCM Input (Bits 15 – 0)**  This register allows input of linear PCM via C-BUS for transcoding. The number format is 2's complement and ranges from \$8000 through \$0000 to \$7FFF (-32768 to 32767). Bit 1 of the CODEC INTERRUPT CONTROL Register (\$81) can be set to a logic 1 to enable interrupts informing a micro-controller when the register should be updated.

# **5.2.1.17 DECODE ADM INPUT Register (\$D8)**

**Decoder ADM Input (Bits 7 – 0)**  This register allows ADM bits to be written into the decoder via C-BUS and is intended for transcoding. Bit 0 of the CODEC INTERRUPT CONTROL Register (\$81) can be set to a logic 1 to enable interrupts informing a micro-controller when the register should be updated. Additionally this register can be loaded with an idle data pattern (\$55 or \$AA) and then selected as the input to the decoder via the DECODER MODE AND SETUP Register (\$D0).

# **5.2.1.18 ENCODER MODE AND SETUP Register (\$E0)**

**Decimation Rate (by 4/8) (Bit 15)**  The encoder PCM filter functions as a decimating lowpass when the encoder is running. PCM values are available in the ENCODE LINEAR PCM OUTPUT Register (\$E6) at the decimation rate. A logic 1 sets the decimation rate to 4 (1/4<sup>th</sup> the bit rate). A logic 0 sets the decimation rate to 8  $(1/8<sup>th</sup>$  the bit rate).

**PCM Input Select (Bits 14 – 13)**  Allows selection of the input to the PCM rate converting filter.

![](_page_29_Picture_207.jpeg)

If PCM filter is interpolating, the encoder can transcode a PCM signal to ADM. If PCM filter is decimating, the encoder will transcode an ADM signal to PCM.

![](_page_29_Picture_208.jpeg)

![](_page_29_Picture_209.jpeg)

#### 2013 CML Microsystems Plc 30 D/649/6

**Local Decoder Output Select (Bits 10 – 9)** 

![](_page_30_Picture_192.jpeg)

# 2013 CML Microsystems Plc 31 D/649/6

# **5.2.1.19 ENCODE ADM CONTROL Register (\$E1)**

**Syllabic Time Constant (Bits 15 – 13)** 

Step size integrator Loss Coefficient: allows selection of syllabic time constant.

![](_page_31_Picture_174.jpeg)

**Dynamic Range for Step Size Integrator** 

Numbers given for maximum and minimum step size are based on 16-bit word length (-32768 to 32767).

**Step Size Integrator (Bits 12 – 10)** 

![](_page_31_Picture_175.jpeg)

**Companding Rule (Bits 9 – 8)** 

This is the number of consecutive ones or zeros that must occur for the step size to be adjusted.

![](_page_31_Picture_176.jpeg)

**Estimator** 

![](_page_32_Picture_211.jpeg)

Allows selection of the estimator integrator time constant.

**Threshold (Bits 15 – 0)**  The number programmed into this register can range from \$0 to \$7FFF (0 to 32767). The equation for the VAD threshold is: Register Value =  $\frac{\text{(Signal Detection Threshold)} \cdot 2}{\text{Total REALP}}$  $=\frac{\text{(Signal Detection Threshold)} \cdot 2^{15}}{\text{(Total R1 R2 A)} \cdot \text{Total R2}}$ 

(DAC Full Scale Reference Voltage)

# **5.2.1.21 ENCODE OFFSET LEVEL Register (\$E3)**

**Encode Offset Input (Bits 15 - 0)**  These bits allow for an offset amount to be directly programmed. This offset amount is useful in trimming out offsets that may occur in the on-chip analog circuitry. The number format is 2's complement and ranges from \$8000 through \$0000 to \$7FFF (-32768 to 32767).

2013 CML Microsystems Plc 33 D/649/6

The equation for the direct offset value is:

(DAC Full Scale Reference Voltage) Register Value =  $\frac{(Offset Voltage) \cdot 2}{(D \cdot 1) \cdot (D \cdot 1) \cdot (D \cdot 2)}$ (Offset Voltage)  $\cdot 2^{18}$ 

For normal Encoder operation this register should be loaded with a small positive constant (eg in the range [2-16]) and bit 8 of the ENCODER MODE AND SETUP Register (\$E0) should be set to logic 1. The programmed offset will be summed with the encoder input signal. If offset compensation is not required, bit 8 of the register \$E0 should be set to logic 0 and the ENCODE OFFSET LEVEL Register should also be set to logic 0.

Offset compensation can be suspended by loading this register with 0 while leaving Bit 8 of register \$E0 true. This holds the current offset estimate constant. The offset estimate can be read out via the ENCODE OFFSET LEVEL OUTPUT Register (\$E5).

#### **5.2.1.22 ENCODE DAC INPUT Register (\$E7)**

**Encode DAC Input (Bits 15 – 0)**  This register allows direct access to the encoder DAC input. The number format is 2's complement and ranges from \$8000 through \$0000 to \$7FFF (-32768 to 32767).

#### **5.2.1.23 ENCODE ADM INPUT TEST Register (\$E8)**

**Encoder ADM Input Test (Bits 7 – 0)**  This register allows ADM bits to be written via C-BUS for transcoding from ADM to PCM. An interrupt can be enabled to inform a micro-controller when the register needs reloading.

## **5.2.2 Read Only Register Description**

#### **5.2.2.1 PROCESSOR STATUS READ Register (\$80)**

Reading this STATUS register clears any pending IRQ. The PCM and ADM data available and data needed flags (bits 5, 4, 1 and 0 respectively) are cleared when the appropriate CBUS register is read (or written), in order to service the IRQ. The VAD detection flags (bits 6 and 2) are constantly updated to indicate the status of voice activity. Any change in state of either flag will cause an IRQ to be generated.

![](_page_34_Picture_222.jpeg)

#### **5.2.2.2 DECODE VAD LEVEL OUTPUT READ Register (\$D4)**

**Decode VAD Level Output (Bits 15 – 0)**  These bits indicate the average amplitude of the envelope of the audio signal. This negative 2's complement number can range from \$0 to \$8000 (0 to -32768 and can be used to assist in calculating an appropriate value to be programmed into the DECODE VAD THRESHOLD Register (\$D2). The equation for the VAD level register value is:

> (DACFull Scale ReferenceVoltage) Register Value =  $\frac{-1}{2}$  (Envelopevoltage level) 2  $=\frac{-1.(\text{Envelope}ot\text{atgelevel})^2}{(2.15 \times 10^{-15} \text{ J/kg} \cdot \text{cm}^2)(1.5 \times 10^{-15} \text{ J/kg} \cdot \text{cm}^2)}$

#### **5.2.2.3 DECODE OFFSET LEVEL OUTPUT READ Register (\$D5)**

**Decode Offset Level Output (Bits 15 – 0)**  These bits indicate offset level as input by the user in register \$D3. The number format is 2's complement and ranges from \$8000 through \$0000 to \$7FFF (-32768 to 32767).

The equation for the offset value is:

(DAC Full Scale Reference Voltage) Register Value =  $\frac{(Offset Voltage) \cdot 2}{P \cdot S \cdot R \cdot R \cdot R \cdot R \cdot R}$ 18

## **5.2.2.4 DECODE LINEAR PCM OUTPUT READ Register (\$D6)**

**Decode Linear PCM Output (Bits 15 – 0)**  This register contains the linear PCM equivalent of the ADM or non-linear PCM input signal. The number format is 2's complement and ranges from \$8000 through \$0000 to \$7FFF (-32768 to 32767). Bit 1 of the CODEC INTERRUPT CONTROL Register (\$81) can be set to a logic 1 to enable interrupts, informing a micro-controller when the register has been updated.

The equation for the PCM register value is:

Register Value =  $\frac{\text{(PCM voltage)} \cdot 2^{15}}{\text{(Det CFT, T.G.)}}$ 

(DAC Full Scale Reference Voltage)

2013 CML Microsystems Plc 35 D/649/6

# **5.2.2.5 DECODE ADM OUTPUT READ Register (\$DA)**

**Decode ADM Output (Bits 7 – 0)**  This register allows ADM bits to be read via C-BUS and is updated every eighth bit. Bit 0 of the CODEC INTERRUPT CONTROL Register (\$81) can be set to a logic 1 to enable interrupts, informing a micro-controller when the register has been updated. When the decoder is set to transcode from PCM to ADM the ADM bits are available via this register.

#### **5.2.2.6 ENCODE VAD LEVEL OUTPUT READ Register (\$E4)**

**Encode VAD Level Output (Bits 15 – 0)**  These bits indicate the average amplitude of the envelope of the audio signal. This negative 2's complement number can range from \$0 to \$8000 (0 to -32768) and can be used to assist in calculating an appropriate value to be programmed into the ENCODE VAD THRESHOLD Register (\$E2).

The equation for the VAD level register value is:

(DACFull Scale ReferenceVoltage) Register Value =  $\frac{-1}{2}$  Envelopevoltagelevel) 2  $=\frac{-1.(\text{Envelope}ot\text{atgelevel})^2}{(2.15 \times 10^{-15} \text{ J/kg})(1.5 \times 10^{-15} \text{ J/kg})}$ 

# **5.2.2.7 ENCODE OFFSET LEVEL OUTPUT READ Register (\$E5)**

**Encode Offset Level Output (Bits 15 – 0)**  These bits indicate the offset level as input by the user in register \$E3, which is dynamically updated if Idle Channel Enhance is enabled. The number format is 2's complement and ranges from \$8000 through \$0000 to \$7FFF (-32768 to 32767). It can be used as an appropriate value to be programmed into the ENCODE OFFSET LEVEL Register (\$E3) if offset compensation will be disabled.

The equation for the offset value is:

Register Value =  $\frac{(Offset Voltage) \cdot 2}{P \cdot 2P \cdot R \cdot 2P \cdot R \cdot R \cdot 2P \cdot R}$  $=\frac{\text{(Offset Voltage)} \cdot 2^{18}}{2 \times 10^{-18} \text{ m} \cdot 10^{-18} \text{ m}^2}$ 

(DAC Full Scale Reference Voltage)

#### **5.2.2.8 ENCODE LINEAR PCM OUTPUT READ Register (\$E6)**

**Encode Linear PCM Output (Bits 15 – 0)**  This register containes the linear PCM equivalent of the encoded ADM signal. The number format is 2's complement and ranges from \$8000 through \$0000 to \$7FFF (-32768 to 32767). Bit 5 of the CODEC INTERRUPT CONTROL Register (\$81) can be set to a logic 1 to enable interrupts, informing a micro-controller when the register has been updated.

The equation for the PCM register value is:

Register Value =  $\frac{(PCM \text{ voltage}) \cdot 2^{15}}{(DAC \text{ Full Scale Reference Voltage})}$ 

#### **5.2.2.9 ENCODE ADM OUTPUT READ Register (\$EA)**

**Encode ADM Output Test (Bits 7 – 0)**  This register allows Encoder ADM bits to be read via C-BUS and is updated every eighth bit. Bit 4 of the CODEC INTERRUPT CONTROL Register (\$81) can be set to a logic 1 to enable interrupts, informing a micro-controller when the register has been updated.

2013 CML Microsystems Plc 36 D/649/6

# **6. Application Notes**

#### **6.1 C-BUS Operation**

Instructions, status and data are transferred between the CMX649 and the host µC over the C-BUS. Instruction and data transfers to and from the CMX649 consist of an Address/Command (A/C) byte followed by either:

- 1. a further instruction or
- 2. 1 or 2 bytes of data (write) or
- 3. 1 or 2 bytes of status or received data reply (read).

The number of data bytes following an A/C byte is dependent on the value of the A/C byte. The most significant bit of the address or data is sent first. The C-BUS SERIAL\_CLOCK input to the CMX649 originates from the host µC.

![](_page_36_Picture_96.jpeg)

**Figure 12 C-BUS Timing Diagram** 

#### **6.2 CODEC Data Interface**

The CMX649 encodes analog audio signals into digital samples, which can be decoded back to analog audio signals. The number of bits per sample is determined by the coding algorithm, which is selected using register \$70, CODEC MODE CONTROL. For the ADM coding algorithm, one bit represents one sample and the linear PCM coding algorithm uses 16 bits to represent one sample. The A-law and μ-law PCM algorithms use 8 bits to represent one sample.

The CMX649 CODEC Data Interface is a digital interface responsible for transferring Rx samples into, and Tx samples out of, the device. The interface can be operated in one of two modes: buffered or unbuffered. This selection is made when using the CODEC MODE CONTROL register to choose the coding algorithm.

In burst mode, also called buffered mode, the device will buffer (hold) a data word (either 8 or 16 bits depending on the operating mode) until it is transferred in for processing or out to the listening device. Burst mode requires two timing signals to be furnished by a controlling device: Sync (STRB Pin 1) and Clock (Rx Clk Pin 12). Tx and Rx clocks are tied together internally for burst mode and are driven by the Rx Clk pin. There is only one sync input.

In non-burst mode the device transfers data one bit at a time and only one timing signal is required, Clock. The clock signal can be produced by the CMX649 or supplied by the controlling device. The Tx and Rx clocks can be separate or the same.

The clocking choices of the CMX649 are very flexible. Please refer to the application note: *CMX649 Operation and Application,* for related detail.

![](_page_37_Picture_174.jpeg)

# **Burst Mode**

![](_page_37_Picture_175.jpeg)

#### 2013 CML Microsystems Plc 38 D/649/6

![](_page_38_Figure_2.jpeg)

# **Figure 13 Burst Interface Timing Diagram for Concatenated Byte Transfers**

Notes for Figure 13:

 $\bullet$  In this example Bit 7 is the most significant bit.

Once started Rx and Tx data bits are continuously streaming so long as the SYNC pulse continues at the PCM sample rate (for PCM modes) or at  $1/8<sup>th</sup>$  the data rate for ADM modes.

Configuration options support some variations of this timing diagram, e.g. data word length, without affecting the timing shown.

• The TX DATA output may be high impedance between burst frames depending on bit 9 of CLK SOURCE CONTROL Register (\$73).

• In ADM mode Data length is 1 bit. Frames are 8 bits in length.

![](_page_38_Figure_10.jpeg)

# **Non-Burst Mode**

![](_page_38_Figure_12.jpeg)

2013 CML Microsystems Plc 39 D/649/6

Notes for Figure 14:

Tx and Rx clocks don't need to be the same rate or phase but the Rx data rate must be at the same rate with which the data was encoded.

Tx and Rx bit clocks can be internally generated by the CMX649 clock generator section, or they can be supplied from external sources.

ADM is the only un-buffered mode possible.

Duty cycle constraints shown for Tx Clk also apply to Rx Clk.

# **6.3 Example CODEC Setups and Application Help**

Below are tabulated some applicable settings for the CLK DIVIDER CONTROL Register (\$72). Note that the minimum crystal frequency that can be used is 8.000 MHz, frequencies below this require the use of an external clock source.

![](_page_39_Picture_133.jpeg)

When selecting divider settings to arrive at a desired bit rate from a given crystal frequency, note that some power savings are realized by selecting a lower divider value in conjunction with a higher prescaler value, thus minimizing the frequency of the prescaler output.

![](_page_40_Picture_153.jpeg)

![](_page_40_Picture_154.jpeg)

#### ADM Codec CMX649

![](_page_41_Picture_163.jpeg)

![](_page_41_Picture_164.jpeg)

```
6.3.1 32kbps ADM with clock and data recovery 
//Initialize device with general reset 
// This powers down everything excluding the xtal oscillator circuit 
\frac{1}{501}//Setup analog section 
// $61 00 filters set for 2.9kHz BW (default after reset) 
// volume=0dB side tone=-21dB and off
$62 $BE
// audio_level=0dB 
$63 $80// power_control everything on (lowest current setting) 
$64 $55 
$65 $55 
// codec mode 
// default $70 $00 ADM unbuffered (continuous bit serial mode) 
// Clock Divider Control 
// using 8.192MHz master clock 
// filter clock prescale/=4 main divider/=8 => 256kHz SCF clock 
// bit clock prescale/=4 encode and decode bit dividers/=1 since constant divider/=64 => 32kbps 
$72 $F9 $C0 
// PLL is off, Bypass PLL Data Filter and Power it Down 
// Internal Decode and Encode clocks from 
// Decode internal clock 
$73 $00 $78 
// setup decoder 
// decimate by 8 
// decode adm input from RX Data 
// adm estimator drives output 
// vad attack tc=4ms and decay tc=128ms 
// normal vad outputs 
$D0 $00 $B8 
// adm encode feedback from comparator, nulling for improved idle – otherwise as decoder 
$E0 $01 $B8
// to enable offset nulling load small positive constant into encoder offset input reg 
$E3 $00 $04 
// adm mode syllabic tc=16ms 
// step size dynamic range 5120/10 
// companding rule = 4 of 4 
// principal tc=0.33ms 
// second order tc=0.083ms 
// encoder zero tc=0.047ms decoder zero tc=N/A 
// decoder zero at 16kHz i.e. bit_rate/2 enabled 
$D1 $6D $51 
$E1 $6D $52 
// vad thresholds ~20mv 
$D2 $02 $00 
$E2 $02 $00 
// prime idle pattern into CBUS ADM source byte regs 
$D8 $AA 
$E8 $AA 
//Scrambler and Descrambler both on, using polynomial $14 (5 bit LFSR) 
$71 $90 $14 
// enable encoder and decoder with no IRQs 
$81 $88 
// note some useful register changes from the above settings<br>// force encoder to output an idle pattern ($E0 $81 $B9) (re
  // force encoder to output an idle pattern ($E0 $81 $B9) (requires $E8 $AA above) 
// force decoder to idle via idling its input ($D0 $88 $B8) (requires $D8 $AA above) 
// force decoder to mute output via direct PCM out ($D0 $86 $B8) (reset default has $D7 $00 $00) 
// turn Scrambler and Descrambler off ($71 $00 $00)
// To Make PLL run with input from data pad 
// internal RX and TX clocks 
// RX data input acting as analog input i.e. data filter and data slicer running 
// ($73 $00 $D2)
```

```
6.3.2 64kbps burst mode Bluetooth Compatible CVSD 
//Initialize device with general reset 
// This powers down everything excluding the xtal oscillator circuit 
\frac{1}{501}//Setup analog section 
// $61 00 filters set for 2.9kHz BW (default after reset) 
// volume=0dB side tone=-21dB and on
$62 $BF
// audio_level=0dB 
$63 $80// power_control everything on (lowest current setting) 
$64 $55 
$65 $55 
// codec mode ADM buffered (burst bytes at 1/8 bit rate mode) 
$70 $01 
// Clock Divider Control 
// with 8.192MHz master clock 
// filter clock prescale/=4 main divider/=8 => 256kHz SCF clock 
// bit clock prescale/=2 main divider/=1 since constant divider/=64 always => 64kHz bit clocks 
$72 $F9 $40 
// PLL is not running 
// internal RX and TX bit clocks both from RX bit clock 
// RX data input acting as digital input for burst mode 
$73 $00 $70 
// setup decoder 
// decimate by 8 
// decode adm input from RX Data 
// adm estimator drives output 
// decode vad driven by adm bits at bit rate 
$D0 $00 $02 
  adm encode feedback from comparator, nulling for improved idle - otherwise as decoder
$E0 $01 $02
// to enable offset nulling load small positive constant into encoder offset input reg 
$E3 $00 $04
// adm mode BT CVSD algorithm 
// syllabic tc=16ms 
// step size dynamic range 1280/10 
// companding rule = 4 of 4 
// principal tc=0.5ms 
// second order tc=N/A 
// zero tc=N/A 
// zero at bit_rate/2 disabled 
$D1 $BD $A0 
$E1 $BD $A0 
// prime idle pattern into CBUS ADM source byte regs 
$D8 $AA 
$E8 $AA 
// enable encoder and decoder with no IRQs 
$81 $88 
// Alternative settings for PCM format using second order ADM algorithm 
// syllabic tc=16ms 
// step size dynamic range 5120/10 
// companding rule = 5 of 5 
// principal tc=0.5ms 
// second order tc=0.0625ms 
// predictor zero tc=0.0234ms for encoder 
// zero at bit_rate/2 enabled for decoder 
//$D1 $AE $A1 
//$E1 $AE $BA 
// codec mode 2=linear PCM buffered (3=uLaw 4=Alaw)
//$70 $02 
// decoder flow for input PCM plus transcode to ADM with offset null and output via VAD output. 
//$D0 $57 $02
//$D3 $00 $04
```
# **7. Performance Specification**

# **7.1 Electrical Performance**

# **7.1.1 Absolute Maximum Ratings**

Exceeding these maximum ratings can result in damage to the device.

![](_page_44_Picture_122.jpeg)

![](_page_44_Picture_123.jpeg)

# **7.1.2 Operating Limits**

Correct operation of the device outside these limits is not implied.

![](_page_44_Picture_124.jpeg)

# **7.1.3 Operating Characteristics**

The following conditions are assumed unless otherwise specified:  $V_{DD}$  = 2.7V to 5.5V at T<sub>AMB</sub> = -40 to +85°C, Audio Test Frequency = 820Hz, Xtal/Clock f<sub>0</sub> = 8.192MHz, Data Rate = 32kbps, Audio reference level (0 dBm0) =  $489$ m $V_{RMS}$ .

![](_page_45_Picture_417.jpeg)

![](_page_46_Picture_78.jpeg)

- **Notes:** 1. Not including any current drawn from the device by external circuits.
	- 2. Input and output signal levels are independent of supply voltage.
	- 3. Passband and stopband corner frequencies are programmable. Specified values are at nominal external clock or crystal frequencies of 4.096, 8.192, 12.288, or 16.384MHz with the master clock divider configured for a divide by 1, 2, 3, or 4 respectively. For other clock or crystal frequencies, passband and stopband corner frequencies must be scaled accordingly.
	- 4. dBmOp units imply the use of a psophometrically weighted filter that is commonly used in voice communication applications per ITU Recommendation G.223.
	- 5. From 400Hz to 3000Hz with a 3.7kHz bandwidth.
	- 6. At  $V_{DD}$  =2.7V, data rate = 64kbps, 2.9kHz bandwidth, offsets compensated and 1<sup>st</sup> order integration only.
	- 7. AUDIO OUT driving a resistive load connected to a voltage source of  $V_{DD}/2$ .

#### If 2 bytes of data tcsoF  $\downarrow$  $H = 2H$  $\bullet$  $\circ$ tcsH  $\leftarrow$  tepH  $\overline{a}$  $\leftarrow$  $\overline{\mathbf{r}}$  $\overline{\mathbf{N}}$  $\overline{\phantom{0}}$  $\tilde{\mathcal{E}}$  $\overline{P}$  $\overline{\phantom{a}}$  $\overline{\phantom{a}}$ 5 რ $\overline{a}$  $\overline{\phantom{0}}$  $\overline{\bullet}$ twx  $\overline{z}$  $\overline{r}$ If 1 byte of data  $\leftrightarrow$ '<br>tcso⊧F  $\downarrow$  $t$  ck  $\frac{1}{2}$  $\frac{1}{2}$  $\circ$  $\circ$  $\frac{1}{2}$  $\begin{array}{c|c} \hline \text{ } & \text{ } \\ \hline \text{ } & \text{ } \\ \hline \end{array}$  $\leftarrow$  $\overline{\mathbf{N}}$ RPLY  $\overline{\phantom{0}}$  $\tilde{\phi}$ **SCIK**  $\overline{4}$  $\overline{+}$ = Level not important or undefined 5  $\overline{5}$  $\bullet$  $\bullet$  $\overline{r}$  $\overline{r}$  $t\log \frac{1}{t}$  $\circ$  $t$  CH  $-$  topH IXVT  $\overline{\phantom{0}}$  $\overline{2}$  $\tilde{\mathcal{O}}$ ╇  $\frac{1}{2}$ 4  $\overline{4}$ **A** tcps  $t$  CK  $\overline{5}$  $\bullet$  $t$  cL  $\ddot{ }$  $\frac{1}{\sqrt{2}}$  $\frac{1}{2}$ CMD ţ  $30\%$  V<sub>DD</sub>  $-$ CSN  $70\%$  V<sub>DD</sub>  $-$ **SCLK** CMD RPLY

# **7.1.3 (continued) C-BUS Timing Diagram**

**Figure 15 C-BUS Timing Diagram** 

<b>C-BUS Timing (see Figure 15)</b>		<b>Notes</b>	Min.	Typ.	Max.	<b>Unit</b>
$t_{CSE}$	CSN Enable to SCIk high time		100			ns
$t_{\text{CSH}}$	Last SCIk high to CSN high time		100			ns
$t_{\text{LOZ}}$	SCIk low to ReplyData Output Enable Time		0.0			ns
t <sub>HIZ</sub>	CSN high to ReplyData high impedance				1.0	μs
t <sub>CSOFF</sub>	CSN high time between transactions		1.0			μs
$t_{\text{NXT}}$	Inter-byte time		200			ns
$t_{CK}$	SCIk cycle time		200			ns
$t_{CH}$	SCIk high time		100			ns
$t_{CL}$	SCIk low time		100			ns
t <sub>CDS</sub>	Command Data setup time		75			ns
$t_{CDH}$	Command Data hold time		25			ns
$t_{RDS}$	Reply Data setup time		50			ns
t <sub>RDH</sub>	Reply Data hold time		0			ns

**Notes:** 1. Depending on the command, 1 or 2 bytes of COMMAND DATA are transmitted to the peripheral MSB (Bit 7) first, LSB (Bit 0) last. REPLY DATA is read from the peripheral MSB (Bit 7) first, LSB (Bit 0) last.

- 2. Data is clocked into the peripheral on the rising SERIAL CLOCK edge.
- 3. Commands are acted upon at the end of each command (rising edge of CSN).
- 4. To allow for differing µC serial interface formats C-BUS compatible ICs are able to work with SERIAL\_CLOCK pulses starting and ending at either polarity.
- 5. Maximum 30pF load on IRQN pin and each C-BUS interface line.

These timings are for the CMX649 and allow faster transfers than the original C-BUS specification.

For codec data interface timing specifications and diagrams please refer to section 6.2.

## **7.2 Packaging**

![](_page_49_Figure_3.jpeg)

![](_page_49_Figure_4.jpeg)

![](_page_49_Figure_5.jpeg)

**Figure 17 20-Lead SOIC Mechanical Outline:** *Order as part no. CMX649D3*

© 2013 CML Microsystems Plc 60 D/649/6 D/649/6

Handling precautions: This product includes input protection, however, precautions should be taken to prevent device damage from electro-static discharge. CML does not assume any responsibility for the use of any circuitry described. No IPR or circuit patent licences are implied. CML reserves the right at any time without notice to change the said circuitry and this product specification. CML has a policy of testing every product shipped using calibrated test equipment to ensure compliance with this product specification. Specific testing of all circuit parameters is not necessarily performed.

![](_page_50_Picture_21.jpeg)