

Additional Resources	Two scripts are available to download from the CML website to assist with speech evaluation on the CMX7262 - Encode2dsk.pes and Decode_from_disk.pes
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1 Introduction

This Application Note describes a set-up for speech evaluation on CML's TWELP vocoder, the CMX7262. This set-up used the ITU-T Coded Speech Databaseⁱ as the source of raw speech samples. Audacityⁱⁱ was selected as the application to play speech samples and also to record speech samples that have been both encoded and then decoded. The SoundBlaster ExtigyTM was chosen as the sound card for generation and recording of the analogue speech signals, due to its high performance, very low noise and ease of use.

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3 History

Version	Changes	Date
1.0	First Release	11/09/2012

4 Advice on Obtaining the Best Quality Speech

Although the end design may use a microphone, it is necessary to use a signal generator that can produce consistent and repeatable voice signals. Traditional evaluation methods using tones, frequency sweeps and audio analysers are not valid for determining the speech performance because the TWELP algorithm is designed for very low bitrates. Speech samples, both clear and with background noise, should be used.

The PC sound card is a readily-available low-cost device to play back speech samples and record the vocoded results. Most personal computers, whether laptops, netbooks or other platforms offer a built-in sound card. Generally, good results can be achieved with these but there are a number of factors to be aware of:

- Most sound cards connect the ground return for signal cables to frame ground.
- The audio connections on sound cards do not usually default to line in, line out. They may also include automatic audio-type sensing, unknown sampling rates and may intentionally distort the signal for effects such as pseudo surround sound.
- There are significant electromagnetic (EM) fields in and around a PC which may be coupled onto the audio signals or directly onto the evaluation hardware.
- Noise is easily coupled into the signal path and degrades test results.
- PCs have fans that produce vibration in the audio band and these can be acoustically coupled onto the audio path.

4.1 Ground connections

Most sound cards connect the ground return for signal cables to frame ground. For desktop machines this means the ground return is most likely connected to Earth. For laptops running on mains power, the ground return may be connected to mains neutral via a low-value capacitor in the switched mode power supply adaptor. This grounding is to reduce EMC but can act as an additional path for noise pickup and can create ground loops¹.

- ✓ For laptops running on battery, the ground return is fully floating and potentially offers a good, low-noise signal source from built-in sound cards.
- ✓ For PCs, a better choice is an external sound card connected via USB. Low-cost external sound cards are available with very high quality reproduction, low-noise recording inputs, flexible interfacing and direct hardware access for the controlling application. Good quality sound cards also have little or no ground loop issues.

4.2 Potential issues with sound cards

If using a built-in sound card it is probably configured for MIC input and headphone output. Automatic sensing is frequently used to detect a device such as a MIC or headset being plugged in. The sound card then applies a sense algorithm to configure the input/output socket appropriately. It may even select a signal processing algorithm to help improve the perceived quality of the sound or produce aural effects sound as pseudo surround sound.

- ✓ Check the device hardware configuration to ensure that the hardware is set for line in and line out.
- ✓ Use differential signal connections where possible as these largely cancel signals picked up on the traces and connecting cables.
- ✓ Disable auto-input sensing if provided.
- ✓ Disable all options for sound enhancement such as 3D effects, surround sound and bass boost.

Most sound cards use a soft codec which utilises the PC hardware processing power. The signal is actually sampled at an undisclosed rate or rates and then rate converted for the application. This indirect

¹ Ground loops have small signal currents flowing in them which generate voltage differences between ground points. There can be many of these ground potentials yielding ac signals at many different frequencies. A cable screen or signal return trace in a signal lead conducts these small currents into a high impedance input where the resulting potential differences are quite large. These potential differences are summed with the wanted signals, adding unwanted noise.

use of sampling rates can create artefacts in the wanted signal. Good quality *external* sound cards usually allow the application to directly control the sound card sample rate. Only the latest versions of Windows, ie Windows 7 or higher, provide optional direct hardware access to a built-in sound card's hardware. Sound sampling requires significant PC resources. In this evaluation the PC controls the PE0002, the USB sound card and the audio application. If the PC has only marginal processing bandwidth then the audio may be degraded by sample latency. The resulting distortion may not be immediately apparent. Use a higher sample rate than required to check that latency is not an issue and then revert to the wanted sample rate.

- ✓ Enable the hardware driver for direct sampling.
- ✓ Preferably use an external sound card
- ✓ Check that artefacts are not present on a built-in sound card by comparing recorded material with the original source.
- ✓ Check that the PC has adequate processing bandwidth for the tasks it is handling.

4.3 Noise arising from EM field coupling

PCs generate significant magnetic and electric fields which can couple into the external hardware by conduction through the connecting leads or by indirect coupling onto leads, components and board traces. In-band noise that is picked up will sum directly onto the speech signal and high-frequency noise that is picked up will alias into in-band noise through the multiple sampling blocks. High frequency noise may also beat together generating lower frequency in-band noise. Small signal noise currents induced on signal leads and hardware traces produce significant noise voltages at high impedance nodes. These then sum into the signal paths.

- ✓ Position all the equipment well away from the PC to reduce the EM field coupling.
- ✓ Use good quality audio connecting leads.
- ✓ Remove external equipment such as scopes or meters when evaluating the speech quality.
- ✓ Use a good quality external USB sound card. These are designed to deal effectively with EM fields, ground loops and lead pick-up.
- ✓ Chokes on connecting leads may help. The choke should be as close as possible to the connector at the PC end.

4.4 Other potential noise sources

The host processor, in this case the PE0002, and the vocoder run high speed digital circuits that generate a lot of high frequency noise. The PE0601-7262 is designed to prevent this noise coupling into the speech circuits on the board. The audio connecting leads, the USB and the power lead are all potential noise sources. The noise may be picked up by the leads, may be conducted from another source via the lead or may be the result of ground potentials.

- ✓ Use a proper, isolated bench power supply to power up the PE0002 and PE0601-7262.
- ✓ Use thick-cored leads to the bench power supply to reduce voltage transients due to current peaks.

4.5 Acoustic noise

Ceramic capacitors and microphones pick up acoustic noise and translate it into electrical signals. The PE0601-7262 uses capacitors that are very good at not coupling acoustic noise but microphones are a significant problem both electrically and acoustically.

- ✓ Place the PE0601-7262 on a vibration-absorbing surface such as a rubberised mat.
- ✓ Do not use a microphone as a sound source.
- ✓ Keep PCs, power supplies and other equipment that contain fans as far away as possible or place them on a different or solid surface.
- ✓ Check acoustic coupling by lifting the PE0601-7262 off the bench surface and running a noise check.

4.6 About common mode and differential mode signals

A differential driver transmits two identical copies of a signal but with one signal inverted. Ambient noise couples more or less equally onto both signals through the connecting cable. The wanted signal is transmitted in differential mode and the noise picked up is known as common mode noise. When connected to a differential receiver, the difference of the signals is obtained. Ideally, this doubles the wanted signal and cancels the common mode signal, thus reducing noise. The cancelling only works if a true differential receiver is provided. Driving a differential signal into an input with different impedances in each input signal path will result in a proportion of the noise remaining uncanceled. This noise will be summed onto the wanted signal.

- ✓ Use differential connections for low-level signal where possible.
- ✓ If connections are not truly differential, better results will probably be obtained if they are configured as single-ended.

5 Equipment Set-up

Equipment used:

- PE0601-7262, Demonstration Kit for CMX7262
- PE0002 Evaluation Kit Interface Card
- SoundBlaster Extigy™ external USB sound card
- Dual isolated power supply
- Audacity application for audio editing and recording
- PC
- Reasonable quality headphones or earphones

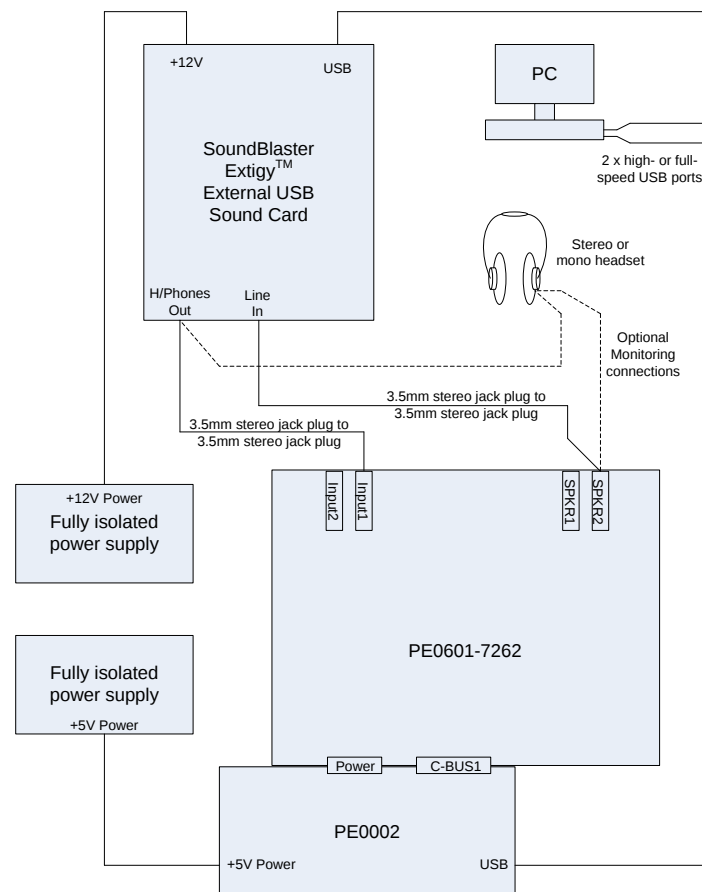


Figure 1. Initial Equipment Set-up

It is necessary to set-up the PE0601-7262 and the audio recording and playback equipment for optimum performance. For convenience, the 3.5mm connections were tested for suitability. It was found that the SoundBlaster Extigy™ produces excellent results in a single-ended configuration and so J1 was used for input. J1 is wired to ANAIN2 on the PE0601-7262. J4 was chosen for the output and this is wired to SPKR2 on the PE0601-7262. The SoundBlaster Extigy™ biases its output to 0V dc and its input to 6V so ac coupling must be used. Suitable ac coupling components are already fitted to the PE0601-7262. See Section 12 for advice on checking and configuring the inputs and outputs for other sound cards.

6 Checking the Noise Floor

It is necessary to start with a low noise floor to get the best performance from the CMX7262. A higher noise floor will cause artefacts in the decoded speech file. For lowest background noise, follow the advice below.

1. Connect the equipment as shown in Figure 1 noting the audio leads are not connected.
2. Power on the PE0002 and the PE0601-7262 then start the PE0002 GUI.
3. Load the FI as described in the PE0002 User Manual.
4. Open the script "Encode2dsk.pes" in a suitable editor.
5. Go to line 7 and enable peak monitoring by changing the constant value to 7.
6. Change line 7 from:

```
peak_select      const 0                ;peak level monitoring. 0=disab..  
to:  
peak_select      const 1                ;peak level monitoring. 0=disab..
```
7. Save the file.
8. Run the script "Encode2dsk.pes" in the PE0002 GUI. See the PE0002 User Manual for loading and running scripts.
9. The CMX7262 is now encoding speech frames and measuring the peak level seen in the frames. The peak level is displayed in the GUI. The values should be less than 30. It is possible to achieve values less than 20. If the peak level is higher than 30, look at the advice on set-up in the previous section.
10. Plug each audio lead, in turn, into the PE0601-7262 input J1 and note the change to the peak level is only a few units. If the level changes by more than 10, the audio lead quality is very low and should be replaced. A well-screened, high quality lead will cause little or no increase in the peak level reading.
11. Connect one audio lead between J1 on the PE0601-7262 and the audio output of the sound card. The peak level may be higher than recorded in 9 but it should not be higher than that recorded in 10. If higher values are seen, check the audio configuration and the sound card. Connect the second audio lead between J4 on the PE0601-7262 and the sound card audio input. If the peak level increases, the sound card is sensitive to a ground loop between the two audio connections. If the screen of the audio lead is accessible, then isolate the screen from the connector's ground and use this connector at J4 on the PE0601-7262. This will break the ground loop without eliminating the screening. Alternatively, connect only one lead, the recording lead or the playback lead, at any one time.
12. Abort the script.

7 Initial Setting for Audacity

Start Audacity and go to preferences. Disable the sound-activated recording mode. On the 'Quality' setting, select a default sample rate of 16000Hz and sample format of 32-bit float. Set the 'Real Time Conversion' to high-quality sinc interpolation and disable the dithering.

Under devices select Windows Direct sound for the Interface. It is advisable to set the 'recording' 'Channels' to '(1) Mono', because playback of a single source recorded on stereo tracks can feel uncomfortable when listened to using headphones. This is a matter of personal preference.

8 Setting the Optimum Levels

When adjusting the gain, set the individual level gain controls in Audacity to 0dB and adjust using the final output control. This gives a good dynamic range from the DAC without compressing signal peaks. Fine adjustments can be made using the individual level gain controls in Audacity. Setting the final gain to maximum may increase the noise on some sound cards depending on the quality of the amplifier used. This is because final gain is in the analogue domain. It may be necessary to compromise by setting the final gain control to the highest level that gives acceptable noise and then add further gain using the individual level gain controls in Audacity. The level controls in Audacity, except the final output gain, change gain in the digital domain.

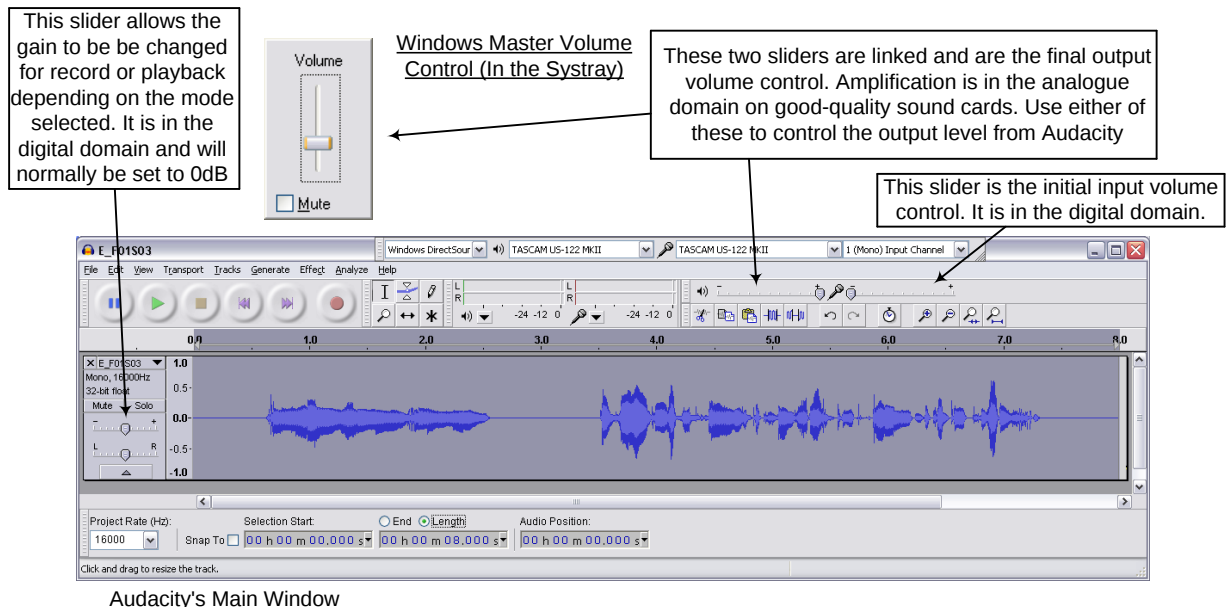


Figure 2. The level controls explained

1. Ensure that the final gain control is set to minimum. On the SoundBlaster Extigy™, this is also linked to the front panel volume control.
2. Connect an audio lead between J1 on the PE0601-7262 and the audio output of the sound card if this is not already in place.
3. Select a speech file and import it into Audacity. Start the speech playback using the looped mode (shift + play) to achieve continuous playback for level setting.
4. Start the script "Encode2dsk.pes" on the PE0002. The peak level mode should still be enabled from the previous section.
5. Increase the playback level using the final gain until any of LEDs 1-4 on the PE0002 are observed to flash. These LEDs act as peak detectors and signal when the corresponding peak level has been exceeded.
6. Increase the playback level until LED 3 illuminates intermittently. LED4 should not illuminate.
7. Observe the full audio sample at the set playback level and make any final adjustment. The maximum dynamic range for the speech sample has not been determined.
8. Abort the script and note the level settings in Audacity for future use.

9 TWELP Encoding a Speech Signal

Set-up Audacity playback levels and the output volume using the settings obtained with the peak detect mode in the previous section.

1. Open the script "Encode2dsk.pes" in a suitable editor.

- Go to line 7 and disable peak monitoring by setting the constant value to 0:

```
peak_select      const 0                ;peak level monitoring.0=disab..
```
- Set the number of frames as required (between 1 and 4):

```
num_frames      word 1                ;1 = 1 frame, 2 = 2 frames, 3 = 3..
```
- Set the FEC as required (0 or 1):

```
FEC             const 1                ; 1 = enabled, 0 = disabled
```
- Set the number of frames for the required recording length:

```
num_packets     word 600                ;set the recording length here
```

Note. 600 sets a recording length of 600 x num_frames x 20ms. If 1 frame transfer is selected, then the recording length will be 12 seconds. The maximum value for num_packets is \$FFFF or 65535.
- Save the file.
- Run the script "Encode2dsk.pes" in the PE0002 GUI. See the PE0002 User Manual for loading and running scripts.
- The script will automatically suspend before starting the vocoder. The PC will display a dialogue requesting the user to start encoding.
- Start the speech sample playing in Audacity.
- Click OK on the PC dialog to start the vocoder.
- Digitally-coded frames will be copied to the PC hard disk until the required recording length is reached when the script will stop.

The file created by the script is saved in the current working directory as "twelp.smp". This is a text file containing the TWELP output samples in created order as hexadecimal bytes followed by a carriage return and line feed (CR+LF).

A file, "vocoder.cfg" is also created to let the decoding script initialise automatically. Using the script pair, configuration only needs to be done in the encoding script.

10 Decoding a TWELP Encoded Speech Signal

When listening to decoded files it is best to use headphones because these help to eliminate ambient noise.

- Connect headphones to J4 on the PE0601-7262.
- Run the script "Decode_from_dsk.pes" in the PE0002 GUI.
- The script will automatically suspend before starting the vocoder. The PC will display a dialogue requesting the user to start decoding.
- Click OK on the PC dialog to start the vocoder.
- The previously encoded file will be decoded and play back through the headphones. The volume can be adjusted to a comfortable level by altering the output attenuation level. This is done by editing line 81 in the script. Line 80 is an example using 5.6dB of attenuation.
- Digitally-coded frames will be fetched from the PC disk until the required recording length is reached when the script and audio playback will stop.

11 Recording a Speech TWELP Encoded/Decoded Sample

It may be helpful to save a copy of the speech sample after it has been TWELP encoded and decoded. This can be saved with the original speech sample to allow direct comparison and this method is described below. The controls in Audacity should be set to 0dB for recording.

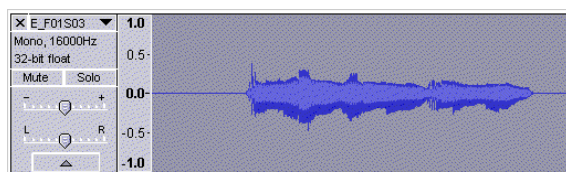


Figure 3. Audacity's Track Control

Note. The Mute and Solo buttons appear above the level and pan sliders.

- Encode a speech sample as described in Section 9.
- Connect an audio lead between J4 on the PE0601-7262 and the sound card audio input.

3. Remove the lead connected to J1 on the PE0601-7262 if this was found to cause additional noise when checked in Section 6.
4. Click on 'Pause' in Audacity's play controls.
5. Click on 'Record' in Audacity's play controls.
6. Another track will open in Audacity below the track of the original speech sample.
7. Run the script "Decode_from_dsk.pes" in the PE0002 GUI and click OK when requested.
8. Note the level on Audacity's Input Level Meter.
9. Adjust the Input Level slider to set the level to the original speech sample.
10. If the level shown on the Input Level Slider is too low, then reduce the attenuator setting in the script "Decode_from_dsk.pes" as described in item 5 of Section 10. Do not add more gain in Audacity unless the attenuator setting of the CMX7262 is at 0dB. This will minimise the SNR of the recording.
11. When the input level is correct, re-run the "Decode_from_dsk.pes" script.
12. When the PC displays the dialogue requesting the user to start decoding, click on 'Pause' in Audacity's play controls. The recording will commence.
13. Click OK on the PC dialog to start the vocoder.
14. When the script has completed, click 'Stop' in Audacity's play controls.
15. Listen to the recording by clicking on 'Solo' in Audacity's track control.
16. Save the speech file as required.

12 Configuring IO for Other Sound Cards or Equipment

This section looks at the impedance and signal levels that can be used with the PE0601-7262 and sound cards or other test equipment. Maximum voltage transfer, not maximum power transfer, is required so the source impedance should be much lower than the load impedance. A useful rule of thumb is to make the source impedance one-tenth of the load impedance. Ground potential differences and signal noise cause noise currents to flow in connecting cables between equipment. When noise currents flow into the input, they generate a voltage proportional to the input impedance and this sums with the desired signal voltage. Therefore lower load impedances are more desirable but this requires a proportional reduction in source impedance. Generally, a 1k Ω source impedance and a 10k Ω load impedance should give good results with signal levels discussed below. Use the minimum amount of equipment necessary for each test in the evaluation.

12.1 Sound card analogue signals

Check the sound card maximum input voltage by injecting a signal from a test set. A sine wave at 1kHz is ideal. Monitor the level in the digital domain using an application such as Audacity. Increase the signal level until it begins to distort. Decrease the gain in the digital domain. If the signal recovers, keep repeating the process until the point at which distortion occurs in the analogue domain. The maximum analogue signal and the corresponding digital level are now known.

Using a resistor decade box and the same test signal, find the input impedance of the sound card input. Connect the signal via the decade box and start at a value around 1k Ω and decrease this to 500 Ω . If the amplitude increases the input impedance is too low and a different sound card should be used. Now increase the resistance and observe the change in signal amplitude. The decade box resistance that causes the amplitude of the signal to drop to half the original is the input impedance. Check the impedance at 300Hz and 3.5kHz to see if it is relatively flat over this range. The input impedance should be around 10k Ω minimum.

Generate a tone using an application such as Audacity. A sine wave at 1kHz and amplitude 0.8 is ideal. Monitor the amplitude using a scope. Increase the analogue volume until the signal begins to distort. If the signal does not distort, increase the gain in the digital domain until distortion occurs. The signal amplitude is the maximum output from the sound card and the corresponding digital setting is now known.

The approximate output impedance can be determined using a resistor decade box. Connect the output signal to ground via the decade box. Start with a value of 1k Ω and increase the value to 2k. If there is an increase in amplitude the output impedance is too high and a different sound card should be used.

Decrease the decade box resistance noting the signal amplitude. The decade box resistance that causes the signal amplitude to fall to half the original level is the output impedance. Check the impedance at 300Hz and 3.5kHz to see if it is relatively flat over this range. The output impedance should be less than 1k.

Check the noise level generated by the sound card or other connected equipment using the method in Section 6. If the noise floor is high and this cannot be resolved using the methods suggested in Section 4 or Section 6 then use a different sound card.

12.2 Identifying the most appropriate analogue input

There are two inputs on the CMX7262, ANAIN1 and ANAIN2.

ANAIN1 is a differential input and if the sound card provides a differential output then use this for optimum SNR. The differential input of the CMX7262 is labelled I_INPUT_P and I_INPUT_N on the PE0601-CMX7262. These pins are accessible on J24:2 and J24:4 respectively. These headers accept only wired connections and must be externally ac coupled.

ANAIN2 is a single-ended input and is accessed by a 3.5mm stereo or mono jack plug fitted at J1. The input signal must be on the tip (the left channel on sound cards). This input is ac coupled on the board so a direct connection to the sound card is possible. A '3.5mm stereo jack to 3.5mm stereo jack' cable is sufficient but use a good-quality one. The ring (the right channel on sound cards) is unconnected. The connector frame is connected to a common analogue ground on the PE601-7262. It is NOT recommended that microphones are used directly in ANAIN2.

A microphone with a suitable, low-noise preamplifier can be used but it must have an unbalanced output for connection to J1. If the preamplifier output is balanced, then both outputs should be externally ac coupled into J24:2 and J24:4. Condenser MICs generally have a built-in low-noise preamplifier but these have high impedance outputs. The high impedance input of J1 makes this input susceptible to noise pickup so a condenser MIC should only be used with a secondary preamplifier module that has a low impedance output.

CD players and other audio devices should be connected to J1 or J24:2 and J24:4 depending on their output configuration. PC CD players have been seen to raise the noise floor, probably due to ground loops via the frame ground and equipment earth. The noise floor tests in Section 6 will confirm the suitability.

For signal generating equipment the most appropriate input should be used; differential where possible, ac coupled where required.

12.3 Identifying the most appropriate analogue output

There are four analogue outputs provided on the CMX7262. These are ANAOUT1, ANAOUT2, SPKR1 and SPKR2.

ANAOUT1 and ANAOUT2 are similar differential outputs. ANAOUT1 connects directly to J19:2 and J19:4. ANAOUT2 connects directly to J19:7 and J19:8. J:19 connections are pin headers and will require ac coupling into a sound card. The output impedance is typically 600Ω but it could be much higher.

SPKR1 is a differential output with a low impedance driver. It is primarily designed for driving into a small speaker and is connected to a 3.5mm socket, J3 on the PE0601-7262. However, the frame of the socket provides the other half of the drive signal and so this output can only be connected to fully differential inputs or fully-floating equipment. The differential signal appears across tip and frame ground. Do not connect ground-screened leads to this socket or the speaker driver could be shorted to ground. SPKR1 is ideal for a headset to monitor the audio output from the CMX7262.

SPKR2 is a single-ended low impedance driver which is ac coupled. It is intended for directly driving a headset and is connected to a 3.5mm socket, J4 on the PE0601-7262. The signal is available on tip and

the ring is floating. Frame ground provides the current return path. The low output, large signal swing and low-distortion makes this connector ideal for recording back into the sound card.

12.4 Signal levels

The maximum peak level of the ADC is +32767, -32768 referenced to V_{analogue} but the available headroom for audio signals depends on the actual signal level, the bias, analogue front-end headroom, the input gain and the algorithm headroom. The available analogue front-end headroom is around 4V differential but the front-end amplifiers will compress this slightly, giving a maximum signal input for a clean peak-peak signal of around 3.5V differential or 1.75V single ended. The recommended maximum peak detector level for encoding is 16384 corresponding to a signal level of +/-1.65V differential or +/-825mV single ended. This assumes that the bias level is 0.5V_{dd} and that the input gain is set 0dB.

For optimal SNR, any input test circuit should have sufficient dynamic range to fit within the allowed peak-peak maximum of 3.3V differential or 1.65V single ended with a bias of 1.65V. Any bias difference should be subtracted from the maximum allowed peak-peak value.

The input impedance of the PE0601-7262 is fixed at 10k Ω minimum for ANAIN1 and 100k Ω for ANAIN2. A source impedance of 1k Ω or less is preferred for the output of the sound card or other test equipment. Because ANAIN2 has such a high impedance, high noise may be seen when using this input. If necessary, the board can be modified to reduce this to around 10k Ω and potentially reduce the noise.

The output signal level of the DAC is close to V_{analogue} but is limited by the headroom of the analogue front end and. Compression occurs in the output buffers within approximately 300mV of rail giving a peak-peak output of 2.7V biased at 1.65V for ANAOUT1 and ANAOUT2. The compression points of the speaker drivers are slightly lower giving an output of 2.55V peak-peak or 5.1V differential for SPKR1 and a peak-peak output of 2.8V for SPKR2 when biased to 1.65V.

The source impedance of ANAOUT1 and ANAOUT2 can be around 2k Ω and they should be buffered or used with an input impedance around 20k Ω or higher. The speaker outputs are designed to drive into low impedances so SPKR1 can be used as a differential source and SPKR2 as a single-ended source.

13 Suitable Sound Samples

This set-up uses speech samples from the ITU T Speech Coded Database, see reference i. This database includes samples of background noise that can be opened in the same Audacity session and played at the same time as a speech track. Other sound samples can be recorded or downloaded from the web. Ensure that samples are not in a lossy compressed format. Files that are typed as MP3 for example, are saved in a compressed format that loses some original content. When compressed or decompressed via TWELP, audible artefacts can be generated.

The CMX7262 allows PCM coded samples to be directly written and read. The referenced scripts do not support this but suitable scripts can be obtained from CML.

14 References

ⁱ ITU-T Recommendation. Supplement 23 to Series P (98-02)
ITU-T Coded Speech Database

ⁱⁱ Audacity V.1.3.1.2 is a free digital audio editing suite available from Sourceforge.net

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