**README**

## LoopBack Project

*Procedure to Open and Run the File :*

- Connect your Microphone to your PC, and connect the Line-out of the PC to the line-in of the board. Connect a headset to the line-out of the board to hear the sound
- Download the zip folder and unzip it in whichever location you prefer. (Say in **Documents**)
- Go to **VMware Workstation 14 Player**>**Windows**>**Start**>**Freescale CodeWarrior IDE**
- Once Freescale CodeWarrior IDE opens, go to **File**>**Open**- Select **Documents**>**Session3_Fu_lab**>**Code**>**Codec_Loopback>Codec_LoopBack** and click Open. The file along with Processor Expert should open.
- Go to Processor Expert tab and click Generate Codec_LoopBack.c . The files should generate in the Files tab on the right-hand-side
- In the right-hand side, Go to User Modules and select Codec_LoopBack.c , and your program should open.
- Go to **Edit>Idm Settings**.. Then under Target Setting Panels, go to **Debugger> Remote Debugging**. Then in Connections Settings, in Connection, in the drop-down menu, select **56800E Simulator.**
- Check the default codec settings of the processor expert by going to the “Project Window”, select the “Processor Expert” tab, double click on “Codec:56858EVM_Codec [Audio_Codec_CS4218]” . The “Bean Inspector” Window should have the ‘Read’ and ‘Write’ methods are enabled.
- Again go to the **Project** tab, click **Debug**. You will enter an **sdm.elf thread window**.
- Click Run two times. The code will be dumped onto the board
- Now speak into the MIC, you should be able to hear through the headset.

## Echo Project

*Procedure to Open and Run the File :*

- Connect your Microphone to your PC, and connect the Line-out of the PC to the line-in of the board. Connect a headset to the line-out of the board to hear the sound
- Download the zip folder and unzip it in whichever location you prefer. (Say in **Documents**)
- Go to **VMware Workstation 14 Player**>**Windows**>**Start**>**Freescale CodeWarrior IDE**
- Once Freescale CodeWarrior IDE opens, go to **File**>**Open**- Select **Documents**>**Session3_Fu_lab**>**Code**>**Echo>Echo** and click Open. The file along with Processor Expert should open.
- Go to Processor Expert tab and click Generate Echo.c . The files should generate in the Files tab on the right-hand-side

- In the right-hand side, Go to User Modules and select Echo.c , and your program should open.
- Go to **Edit>Idm Settings**.. Then under Target Setting Panels, go to **Debugger> Remote Debugging**. Then in Connections Settings, in Connection, in the drop-down menu, select **56800E Simulator.**
- Check the default codec settings of the processor expert by going to  the “Project Window”, select the “Processor Expert” tab, double click on “Codec:56858EVM_Codec [Audio_Codec_CS4218]” . The “Bean Inspector” Window should have the  ‘Read’ and ‘Write’ methods are enabled.
- Vary the frequency of the codec by setting the switch S5 to appropriate sampling frequencies.
- Go to the **Project** tab, click **Debug**. You will enter an **sdm.elf thread window**.
- Click Run two times. The code will be dumped onto the board
- Now speak into the MIC, you should be able to hear through the headset. If you vary the attenuation or the sampling frequency(through the switch S5) or the Delay time (through Buffer Size), you can hear the echo across the headset, sometimes faster or slower, depending on your variations.

**Filter Project**

*Procedure to Open and Run the File :*

- Connect wires from the line-in of the board to the Function Generator and the line-out of the board to an Oscilloscope.
- Download the zip folder and unzip it in whichever location you prefer. (Say in **Documents**)
- Go to **VMware Workstation 14 Player**>**Windows**>**Start**>**Freescale CodeWarrior IDE**
- Once Freescale CodeWarrior IDE opens, go to **File**>**Open**- Select **Documents**>**Session3_Fu_lab**>**Code**>**Filter1>Filter1** and click Open. The file along with Processor Expert should open.
- Go to Processor Expert tab and click Generate Filter1.c . The files should generate in the Files tab on the right-hand-side
- In the right-hand side, Go to User Modules and select Filter1.c , and your program should open.
- Go to **Edit>Idm Settings**.. Then under Target Setting Panels, go to **Debugger> Remote Debugging**. Then in Connections Settings, in Connection, in the drop-down menu, select **56800E Simulator.**
- Check the default codec settings of the processor expert by going to  the “Project Window”, select the “Processor Expert” tab, double click on “Codec:56858EVM_Codec [Audio_Codec_CS4218]” . The “Bean Inspector” Window you must make sure, under Codec Mode, choose **MONO** instead of STEREO to avoid doubling your cutoff frequency.
- Set the sampling frequency by changing the appropriate **S5 switches**. ( Say, OFF-OFF-OFF for 8KHz of sampling frequency)

- Now, in Windows10, open **MATLAB** and go to invoke the **FDATOOL** box to export the filter coefficients of the FIR lowpass filter. Enter order, Passband frequency (Fpass), stopband frequency (Fstop), Sampling frequency (Fs), Weight for each of the pass and stop bands, choose **FIR Lowpass Equiripple** filter, and click Design Filter. After it is done, go the **File> Export> Workspace**.( In the Project, the values are for Fpass=1700Hz, FStop=2000Hz)
- In the Workspace, since we have to convert the coefficients into Q15 formats, multiply each of them by 2^15 and round off to the nearest integer. Then copy all these values in h[size] in Filter1.c.
- Go to the **Project** tab, click **Debug**. You will enter an **sdm.elf thread window**.
- Click Run two times. The code will be dumped onto the board
- Switch on both the Function Generator and the Oscilloscope, and set the Function generator to a Sine wave of amplitude 200mV and frequency 200 Hz. Set the Oscilloscope to measure Pk-Pk amplitude.
- Click Autoset. You should be able to see the Sine wave.
- Vary your frequency between 200Hz till 4000 Hz to notice how the amplitude changes. Baring one or two fluctuations in the amplitude at frequencies below the passband frequency, the amplitude will start to decrease abruptly at the cutoff and will become zero at the stopband frequency.

**Interrupts to implement Loopback and Filter Project**

*Procedure to Open and Run the File :*

- Connect the Line-out of the PC to the line-in of the board. Connect a headset to the line-out of the board to hear the sound
- Download the zip folder and unzip it in whichever location you prefer. (Say in **Documents**)
- Go to **VMware Workstation 14 Player**>**Windows**>**Start**>**Freescale CodeWarrior IDE**
- Once Freescale CodeWarrior IDE opens, go to **File**>**Open**- Select **Documents**>**Session3_Fu_lab**>**Code**>**Filter>Filter** and click Open. The file along with Processor Expert should open.
- Go to Processor Expert tab and click Generate Filter.c . The files should generate in the Files tab on the right-hand-side
- In the right-hand side, Go to User Modules and select Filter.c , and your program should open.
- Go to **Edit>Idm Settings**.. Then under Target Setting Panels, go to **Debugger> Remote Debugging**. Then in Connections Settings, in Connection, in the drop-down menu, select **56800E Simulator.**
- Check the default codec settings of the processor expert by going to the “Project Window”, select the “Processor Expert” tab, double click on “Codec:56858EVM_Codec [Audio_Codec_CS4218]” . The “Bean Inspector” Window you must make sure, under Codec Mode, choose MONO instead of STEREO to avoid doubling your cutoff frequency.

- Set the sampling frequency by changing the appropriate S5 switches. ( Say, OFF-OFF-OFF for 8KHz of sampling frequency)
- Now, in Windows 10, open MATLAB and go to invoke the FDATOOL box to export the filter coefficients of the FIR lowpass filter. Enter order, Passband frequency (Fpass), stopband frequency (Fstop), Sampling frequency (Fs), Weight for each of the pass and stop bands, choose FIR Lowpass Equiripple filter, and click Design Filter. After it is done, go the File> Export> Workspace.( In the Project, the values are for Fpass=1500Hz, FStop=2000Hz)
- In the Workspace, since we have to convert the coefficients into Q15 formats, multiply each of them by 2^15 and round off to the nearest integer. Then copy all these values in h[size] in Filter1.c.
- Go to the **Project** tab, click **Debug**. You will enter an **sdm.elf thread window**.
- Click Run two times. The code will be dumped onto the board
- Play music on the PC, and hear it each time after pressing the IRQA, and then pressing the IRQB. You should be able to hear a filtered sound from the former and a loopback from the latter.

**Implementing the Filter function in Hybrid Project**

- Connect wires from the line-in of the board to the Function Generator and the line-out of the board to an Oscilloscope.
- Download the zip folder and unzip it in whichever location you prefer. (Say in **Documents**)
- Go to **VMware Workstation 14 Player**>**Windows**>**Start**>**Freescale CodeWarrior IDE**
- Once Freescale CodeWarrior IDE opens, go to **File**>**Open**- Select **Documents**>**Session3_Fu_lab**>**Code**>**HFilter>HFilter** and click Open. The file along with Processor Expert should open.
- Go to Processor Expert tab and click Generate HFilter.c . The files should generate in the Files tab on the right-hand-side
- In the right-hand side, Go to User Modules and select HFilter.c , and your program should open.
- You should be able to see HFilter being set as Hybrid assembly.
- Go to **Edit>Idm Settings**.. Then under Target Setting Panels, go to **Debugger> Remote Debugging**. Then in Connections Settings, in Connection, in the drop-down menu, select **56800E Simulator.**
- Check the default codec settings of the processor expert by going to the “Project Window”, select the “Processor Expert” tab, double click on “Codec:56858EVM_Codec [Audio_Codec_CS4218]” . The “Bean Inspector” Window you must make sure, under Codec Mode, choose MONO instead of STEREO to avoid doubling your cutoff frequency.
- Set the sampling frequency by changing the appropriate S5 switches. ( Say, OFF-OFF-OFF for 8KHz of sampling frequency)

- Now, in Windows 10, open MATLAB and go to invoke the FDATOOL box to export the filter coefficients of the FIR lowpass filter.  Enter order, Passband frequency (Fpass), stopband frequency (Fstop), Sampling frequency (Fs), Weight for each of the pass and stop bands, choose FIR Lowpass Equiripple filter, and click Design Filter. After it is done, go the File> Export> Workspace.
- In the Workspace, since we have to convert the coefficients into Q15 formats, multiply each of them by 2^15 and round off to the nearest integer. Then copy all these values in h[size] in Filter1.c. ( In the Project, the values are for Fpass=1500Hz, FStop=2000Hz)
- Go to the **Project** tab, click **Debug**. You will enter an **sdm.elf thread window**.
- Click Run two times. The code will be dumped onto the board
- Switch on both the Function Generator and the Oscilloscope, and set the Function generator to a Sine wave of amplitude 200mV and frequency 200 Hz. Set the Oscilloscope to measure Pk-Pk amplitude.
- Click Autoset. You should be able to see the Sine wave.
- Vary your frequency between 200Hz till 4000 Hz to notice how the amplitude changes. Baring one or two fluctuations in the amplitude at frequencies below the passband frequency, the amplitude will start to decrease abruptly at the cutoff and will become zero at the stopband frequency.