BM83 MSPK V2.0 Design Limitation

D01R05

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2 DESIGN LIMITATIONS

2.1 Concert / Stereo Mode

1. Concert/Stereo mode requires BM83 to be configured as I2S master role.
   Concert/Stereo mode application requires all speakers play the audio under timing synchronized, to make audio synchronization for all speakers, user have to configure BM83 as I2S master role, for both Concert/Stereo master speaker and slave speaker.
   If user don't require Concert or Stereo mode, BM83 can be configured as I2S slave.
2. Slave Speaker may delay (2-3 seconds) to start output audio after grouping while Master Speaker is connecting with PC / NB.
   If the remote device is PC/NB, considering it may connect to keyboard, mouse and other more BT devices BM83 will not force to be BT Master role. So, the performance of RF bandwidth management will be bit worse than the connection with phones. In this case, the commands of sync process cannot send to slaves on time. Slaves spend few times to sense this situation and try to recover the audio sync. It is reasonable and expected result based on our design.
3. Longer A2DP latency is required in Concert / Stereo Mode
   For Single Mode A2DP playing, default 240ms latency is applied and it is configurable in Configuration Tool. Same default latency is applied on Stereo Mode as well. Meanwhile, in Concert Mode, media packet use relay strategy to transmit media packet from Master speaker to Slave speaker, so the transmit throughput is affected by interference more. A2DP needs 100ms extra latency to have better audio performance.
   Besides, A2DP SBC codec usually takes larger packet size, the packet lost has higher penalty in audio sequency, we add 60ms extra latency if connected device use SBC codec.
   Despite extra A2DP latency could be used. Total latency larger than 400ms is not allowed.
4. Line-in latency in Concert / Stereo Mode recommendation.
   Line-in latency is also configurable in Configuration Tool. Default 80ms for Single mode Line-in loopback is used and shorter than 80ms is acceptable. Regarding to Concert Mode / Stereo Mode, the latency is max to 80ms before firmware v1.3.4 and can be max to 160ms later than v1.3.4. The audio packet loss may be due to environment and interference. With the shorter line-in latency, it may not overcome the loss and cause the audio quality issue.
5. If enable voice prompt at Slave Speaker under Concert / Stere Mode, please note that it might cause audio unsync after Slave Speaker rings voice prompts.
6. The AAC payload size in Android phones is larger than in iPhone. We can image that Android phone will take higher traffics than iPhone when play the same music encoded with AAC. This result in the audio performance may not be good enough at BT and Wi-Fi coexistence in some Android phone due to RF transport control.

2.2 LDAC related

1. LDAC is not supported with internal codec. BM83 Internal codec only supports 20bits, but LDAC specification requires the codec to support more than 24-bits.
2. BM83 try to be BT slave role during LDAC playback.

Some mobiles phone (e. g. Samsung S8) do not receive ACK properly when they are BT slave role and BM83 is the BT master role during LDAC playback. This will reduce throughput and make sound breaks happen because LDAC transmit a larger media packet. To have better performance in this circumstance, BM83 try to be BT slave automatically after LDAC codec was selected (by role switch).

3. The multiple link limitation in LDAC playback
BM83 try to be BT Slave role in the link to have a better performance if LDAC was selected as A2DP playing codec. For multi-link usage, while second ACL link was created, it would become a scatter net (no matter BM83 be another BT slave role or master role) and result in maximum num of A2DP link will be only two.

2.3 Audio Break related

1. Audio Break occurred while Android Phone playing music and trigger BLE connection.
We observed Android phone will change Connection Interval as 7.5 ms to query Service Table. It will cause lower throughput of A2DP streaming. On the other hands, since RF bandwidth will be occupied by A2DP streaming, lower BLE throughput is expected. At MSPK2 design, when BT and BLE coexisted, the BLE link is for signal exchange and not suitable for large data transmission.
2. On Concert mode at slave side, there may be some media payloads missing received due to
    - limited retransmission from Concert master encounter environment interference exists.
    - RF resource was conflict in the coexistence of BLE and Concert mode.

   In such case PLC (Packet loss concealment, silence packet will be used) is introduced to compensate missed payload to aids stream synchronization between master and slave (like high light part on below WAV record), But it will result in soft audio break heard no matter audio source is Line-in or A2DP.

2.4 BLE, BT, SCO, Concert/Stereo Mode coexistence

1. Coexistence of BLE and SCO link in OTA DSP tuning
While performing voice DSP tuning via BLE, SCO link is created for transmitting/receiving SCO voice and the voice waveform was captured to measure DSP parameter effect. We suggest, customers do not perform tuning and transmit/receive SCO voice simultaneously if BLE master and SCO receiver are not in the same device. Otherwise, BLE disconnection might be observed.
2. When BLE coexist with SCO, it is possible to observe LE link lost if remote LE master device send Link Layer command with specific instance (connection update/channel update) frequently.
    - Current design will try to reserve more bandwidth for SCO quality.
    - If the remote device is iPhone, this issue will happen more frequently.
3. Due to RF resource is shared for all activities, when BLE is coexisting with CSB and BT, BLE should lower down priority and keep an acceptable response time. Otherwise, BT/CSB audio break could be heard or even BLE connection may be lost. A proper connection interval is needed to balance BLE, BT and CSB RF resource sharing. In such consideration, we provide a default setting (min=90ms, max=100ms) in configuration tool.

This connection interval parameter will be requested to be updated after MBA has been connected. If customer develop their own APP, then they need to find a suitable timing to call following API which used to request update connection parameter. API details could be found in SDK API document.

```
uint16_t BLE_L2CAP_ConnParaUpdateRequest (uint16_t connHandle, uint16_t intervalMin, uint16_t intervalMax, uint16_t latency, uint16_t timeout);
```

2.5 Remote Connected Device

1. Sony Z5

   During BM83 A2DP playback from SONY Z5 phone, MBA is started on Sony Z5 and use it to connect to this BM83, there will be 1-second audio break during BLE connection.
   Sony Z5 will be updating the BLE connection interval to 7.5ms when it tries to discover service table from BM83. At this moment, the mobile phone will be busy so that the bandwidth of the media packet will be impacted, and the throughput will be dropped down, resulting in audio break.

2. iPhone series

   iPhone do not display battery level for Bluetooth accessory.
   We observed iPhone do not display battery level if CoD (class of device) of BM83 was setting to 0x240414 (for minor device class is 0x14 Loudspeaker). But iPhone would display battery level if CoD of BM83 was setting to 0x240404 (for minor device class is 0x04 Wearable Headset Device). Therefore, we choose the latter one as default setting.

3. PC/Notebook

   PC/ Notebook is normally a Bluetooth master role when connected with Bluetooth HID device, such as mouse and keyboard. If BM83 were a master role in connection with PC, it will make the Bluetooth topology form a scatternet. PC will not transmit media payloads smoothly resulting in serious audio break. To avoid this situation, BM83 will try to be a slave role. For BM83 multi-link usage, while second ACL link was created, it would become a scatter net (no matter BM83 be another BT slave role or master role) and result in maximum number of A2DP link will be only to two.

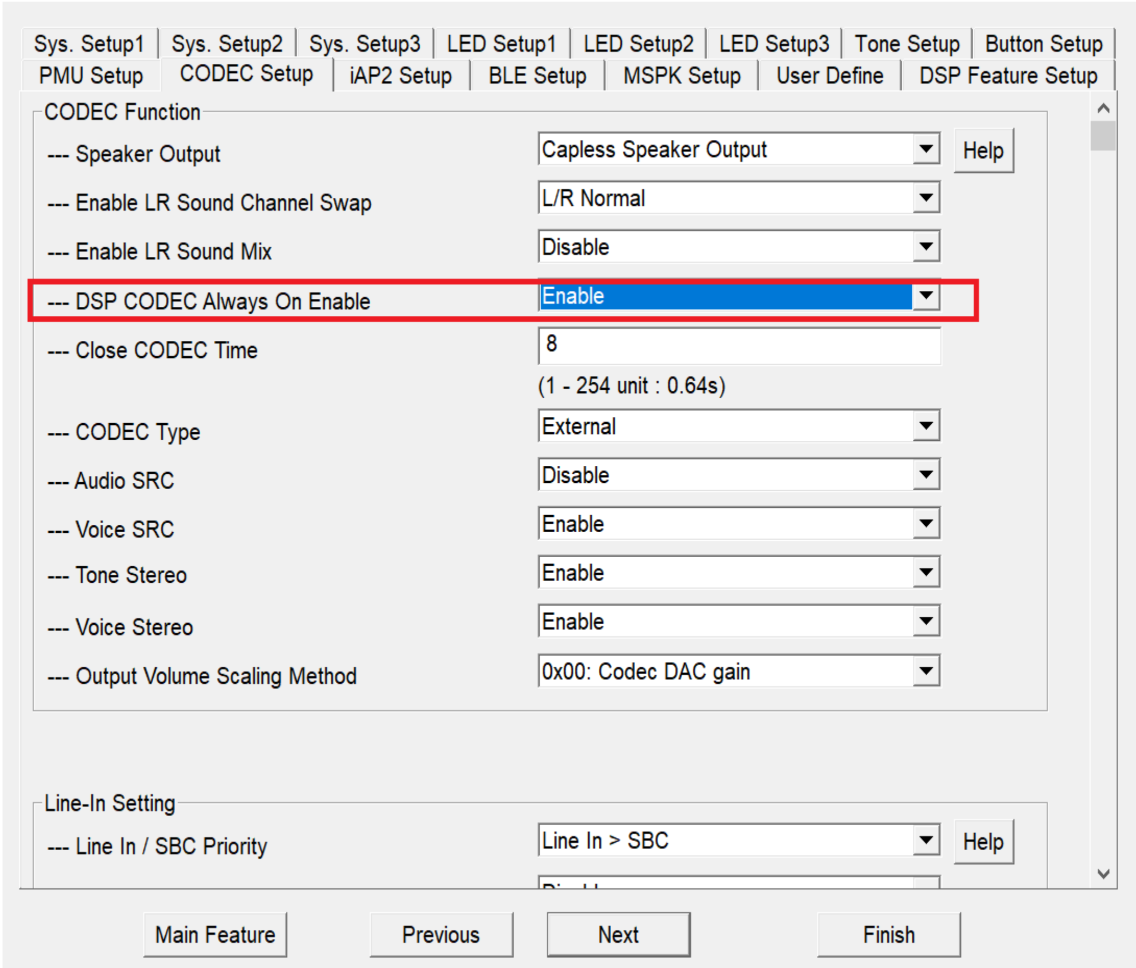
2.6 DSP Audio Clock Source related

1. Description:

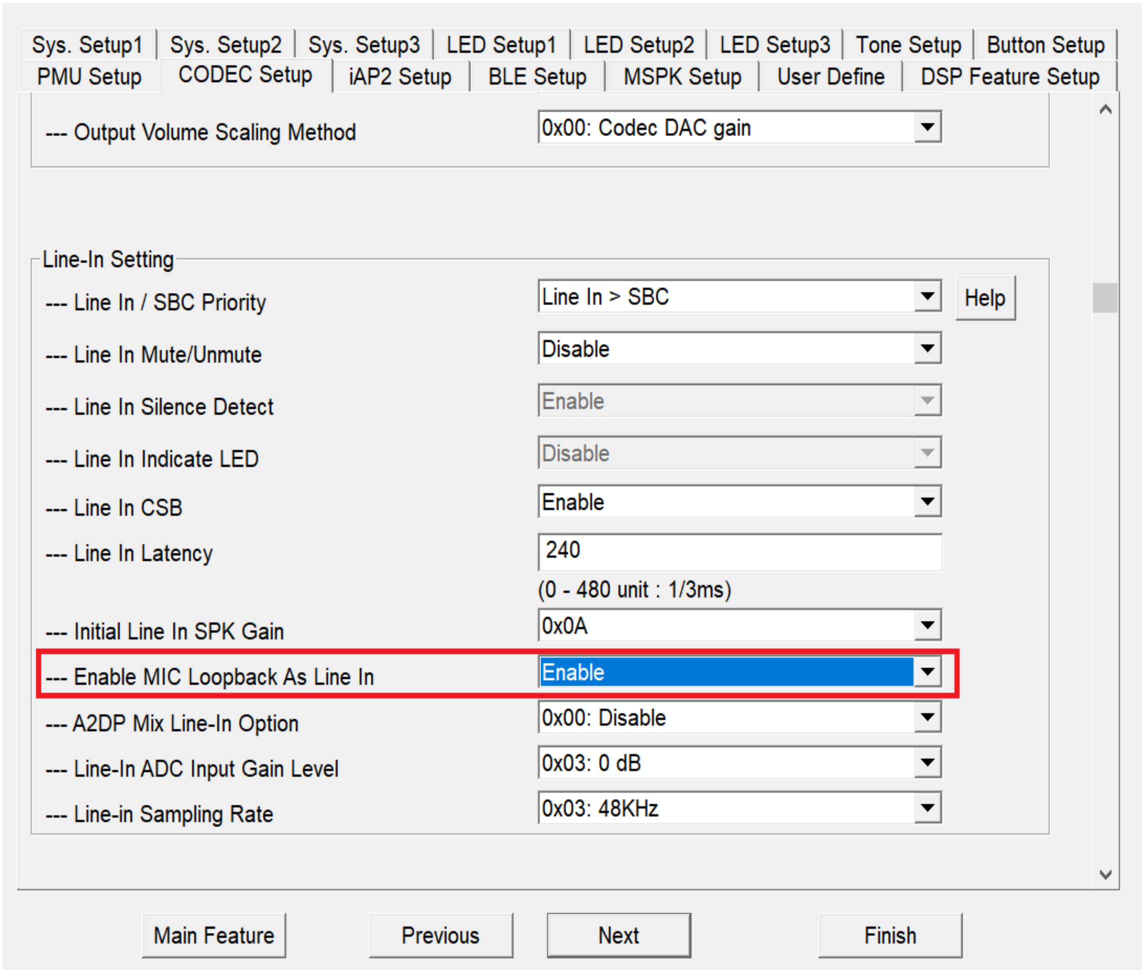
   When the feature “codec always on” is enabled, BT FW avoids discontinuous I2S signal, including BFS and BCLK, by not resetting all the parameters of I2S in DSP during codec initialization, which leads to the selection of the fixed 8MHz ADC clock source for LINE-IN applications in DSP. However, digital microphone (DMIC) cannot be used with fixed 8MHz clock source in the audio subsystem. Customers can only choose analog microphone (AMIC) for the LINE-in loopback application if the signal source is from microphone and the feature “codec always on” is enabled.

2. Usage of UI Config Tool:

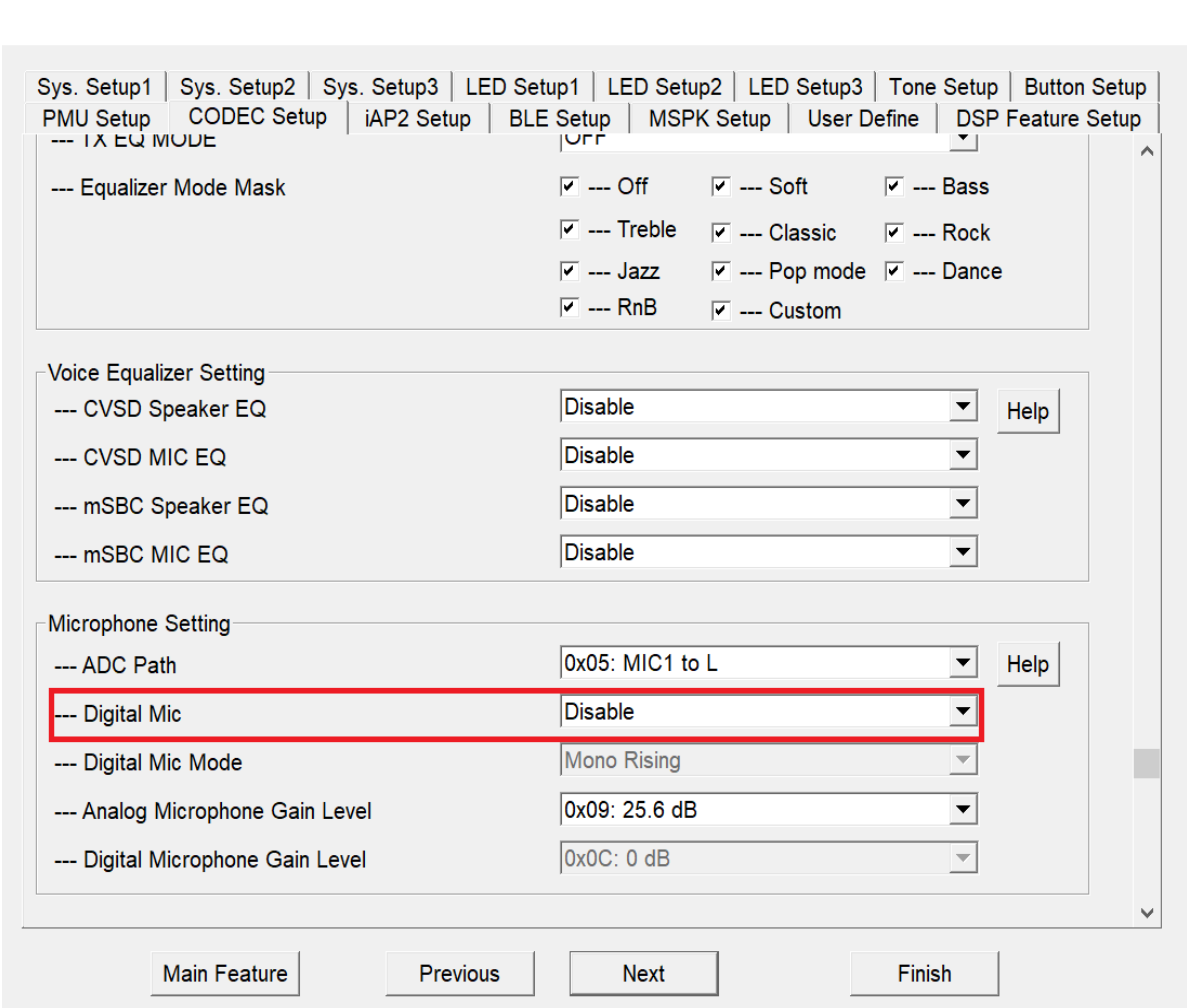
   If the “ DSP CODEC Always On Enable” was enabled and “Enable MIC Loopback As Line In”, then the customer can only use AMIC as ADC input. The steps are shown in the below screenshots from Config Tool.
   A.


Sys. Setup1
Sys. Setup2
Sys. Setup3
LED Setup1
LED Setup2
LED Setup3
Tone Setup
Button Setup
PMU Setup
CODEC Setup
iAP2 Setup
BLE Setup
MSPK Setup
User Define
DSP Feature Setup
CODEC Function
--- Speaker Output
Capless Speaker Output
Help
--- Enable LR Sound Channel Swap
L/R Normal
--- Enable LR Sound Mix
Disable
--- DSP CODEC Always On Enable
Enable
--- Close CODEC Time
8
(1 - 254 unit : 0.64s)
--- CODEC Type
External
--- Audio SRC
Disable
--- Voice SRC
Enable
--- Tone Stereo
Enable
--- Voice Stereo
Enable
--- Output Volume Scaling Method
0x00: Codec DAC gain
Line-In Setting
--- Line In / SBC Priority
Line In > SBC
Help
Main Feature
Previous
Next
Finish

B.


Sys. Setup1
Sys. Setup2
Sys. Setup3
LED Setup1
LED Setup2
LED Setup3
Tone Setup
Button Setup
PMU Setup
CODEC Setup
iAP2 Setup
BLE Setup
MSPK Setup
User Define
DSP Feature Setup
--- Output Volume Scaling Method
0x00: Codec DAC gain
Line-In Setting
--- Line In / SBC Priority
Line In > SBC
Help
--- Line In Mute/Unmute
Disable
--- Line In Silence Detect
Enable
--- Line In Indicate LED
Disable
--- Line In CSB
Enable
--- Line In Latency
240
(0 - 480 unit : 1/3ms)
--- Initial Line In SPK Gain
0x0A
--- Enable MIC Loopback As Line In
Enable
--- A2DP Mix Line-In Option
0x00: Disable
--- Line-In ADC Input Gain Level
0x03: 0 dB
--- Line-in Sampling Rate
0x03: 48KHz
Main Feature
Previous
Next
Finish

C.



2.7 Miscellaneous

1. A2DP / Aux-in source switch in A2DP suppress Aux-in case

   While configured to A2DP has higher priority than line-in, switching audio source to line-in from A2DP by MBA could not make line-in music play immediately. This is because most phone will transmit silence A2DP music after paused A2DP. Due to configured to A2DP has higher priority than line-in, the silence music is still decoding but unable heard, this make line-in music delay to play until A2DP silence music is really terminated. The delay time is various by different phone brand. Sometimes it could be lonest to more than 10 seconds.

3 REVISION HISTORY

Version	Date	History
R1	2021/04/09	Initiate this document
R2	2021/04/16	revised
R3	2021/06/15	Add 2.1 Concert / Stereo Mode Add 2.5 Remote connected device
R4	2021/06/16	Reorganize 2.5 and 2.6
R5	2021/06/23	Reorganize 2.6 and 2.7 Add 2.6 DSP Audio Clock Source related

