FreeSpace® Integrated Zone Amplifiers

Theory of Operation

1. Introduction and system overview
The FreeSpace Integrated Zone Amplifiers are a new power amplifier family that are used to drive Bose® existing DS16 and DS40 speaker systems with individual EQ settings to match.

These amplifiers are available in four variants.
- IZA 250-LZ (Lo Z Main unit): 2 x 50W @4Ohm, 1 RU height x 1/2 width
- ZA 250-LZ (Lo Z Slave unit): 2 x 50W @4Ohm, 1 RU height x 1/2 width
- IZA 190-LZ (Hi Z Main unit): 1 x 90W @ 70/100V, 1 RU height x 1/2 width
- ZA 190-LZ (Hi Z Slave unit): 1 x 90W @ 70/100V, 1 RU height x 1/2 width

2. Features – IZA-250 Master Unit
<u>Five input channels</u>.
- Inputs 1 and 2: Two pairs of RCA connectors on the rear, with input trim
- Input 4: One mic/line combo XLR on the front, with input trim
- Input 5: One 1/8” input on the front (fixed gain)
- Input 3: One 4-pin euroblock for mic/line level page that accommodates page from telephone system and PTT, with input trim

<u>Output channels</u>:
- 2 x 50W (4 ohm) / 2 x ~35W 8 ohm
- One set of tone controls (bass and treble) for the pair of outputs
- Auxiliary output – left and right

<u>Priority and Operation of Inputs</u>
- Source Select Switches between Input 1 and 2
- If source is plugged into 1/8” (Input 5) on the front, it overrides Inputs 1 and 2
- When Input 3 is activated, it takes priority over all inputs and ducks the sources.
- When Input 4 is activated, the front mic/line input can either be mixed without ducking or can override the sources and duck the sources. Requires auto detect for ducking.
- Switch for on/off option of priority inputs 4 and 3, being subject to master volume or not

<u>User Control</u>
- Volume control and source select on the front of the unit
- If remote control is connected, it overrides source select and master volume on the front of the unit. The remote would include two position source select and volume control.

<u>Mounting Options and Size of Main amp and Slave</u>
- Mounting on table (rubber feet)
- Mounting in rack (small as depth as possible) with accessory rack mount kit
- 1 RU high
- Two can be mounted side by side in a rack 1 RU high by full width with accessory kit
- Can be mounted on the wall with accessory kit

<u>Operating Modes</u>
- Two modes, mono and stereo

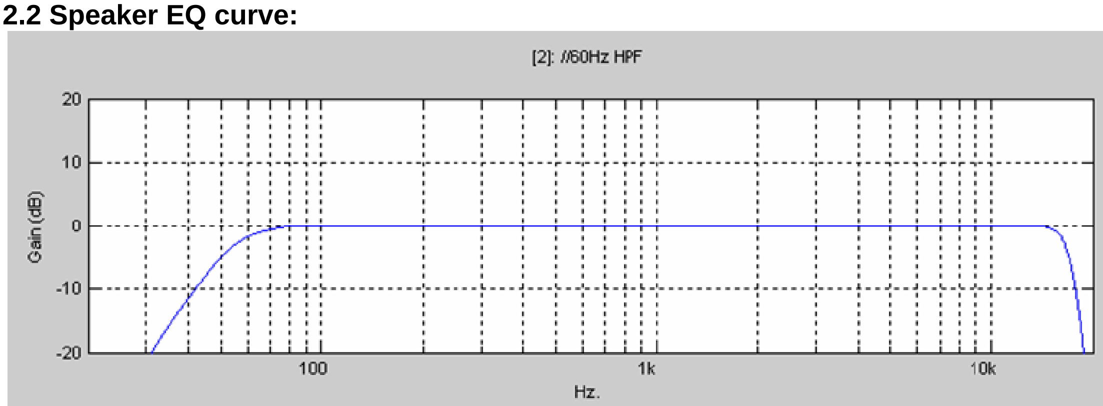
EQ
- DS16, DS40, or high pass filter
- Dynamic EQ
- OptiVoice EQ on priority inputs 4 and 5

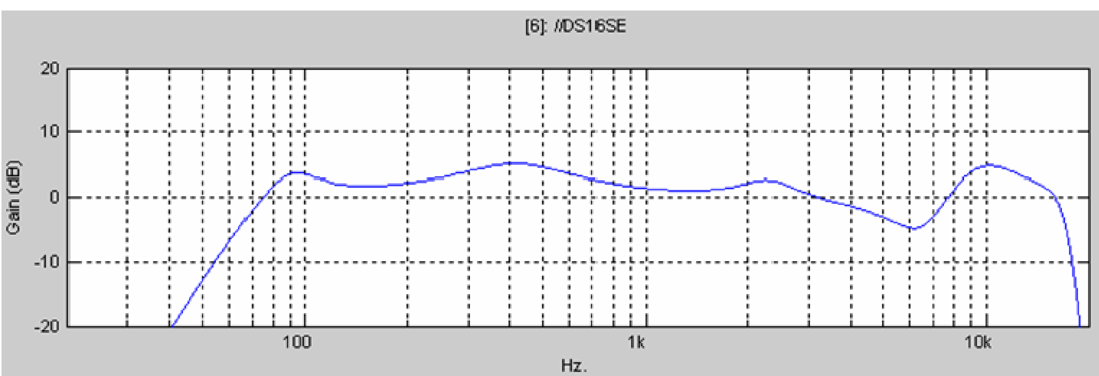
Other Design Items
- Universal Power supply
- Convection cooled – no fan
- Nominal 4 ohms for minimum impedance for low impedance outputs
- Contact closure connection on rear to mute amplifier
- Power LED on front panel
- One clip LED on front, which lights red when a source is clipping

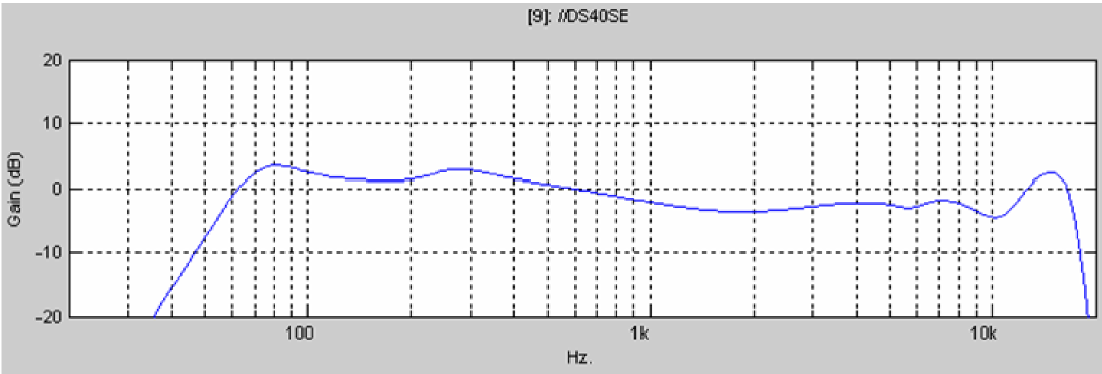
Frequency Response
- Low Impedance: 40 Hz – 18kHz +/- 1dB
- High Impedance: 60Hz – 18kHz +/- 1dB

Input Gain ranges:
CH1, 2 Gain range -20 to +20 dBG
CH3 0 dB fixed
Ch4, 5 0 to 50 dBG

2.2 Speaker EQ curve:

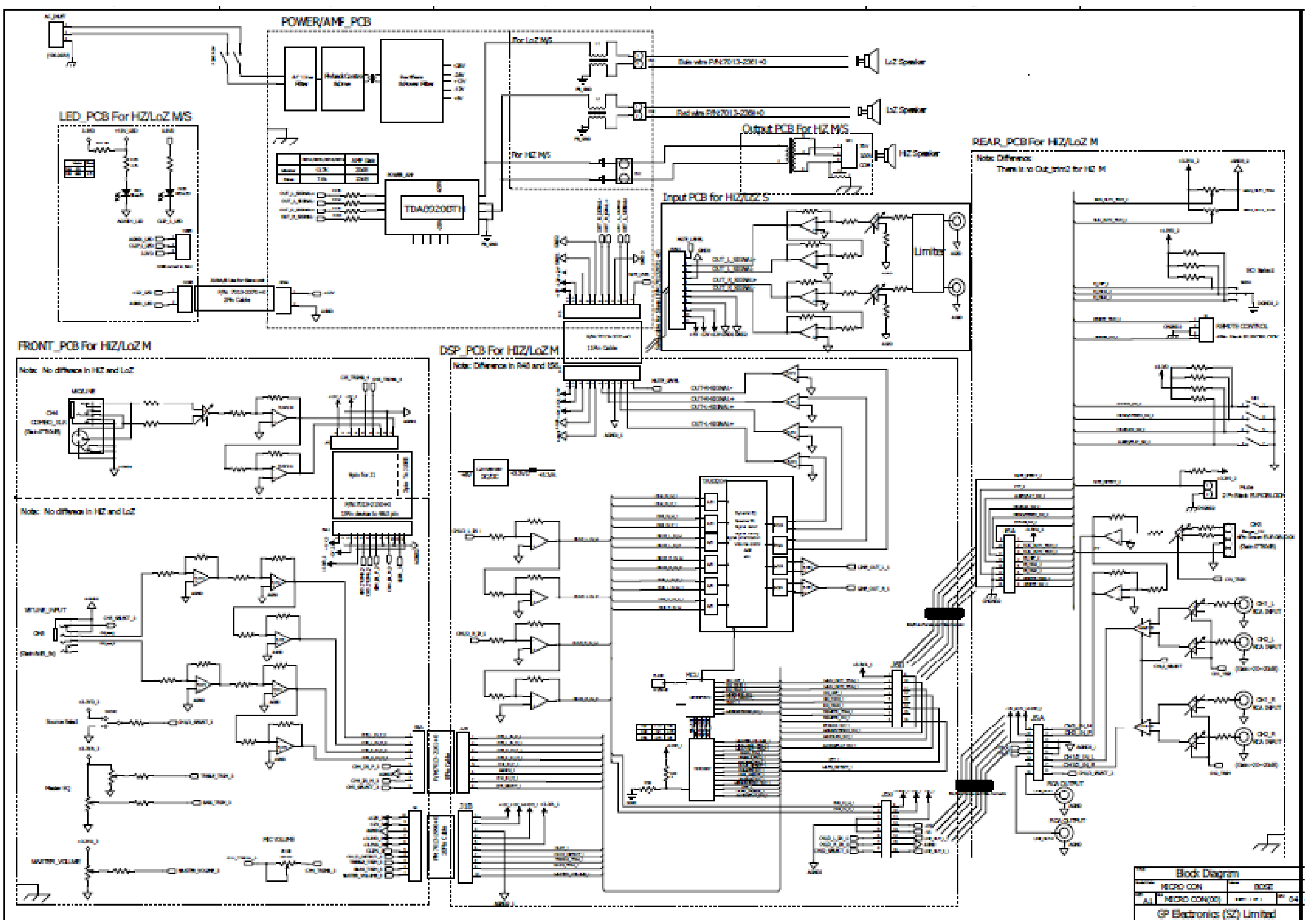




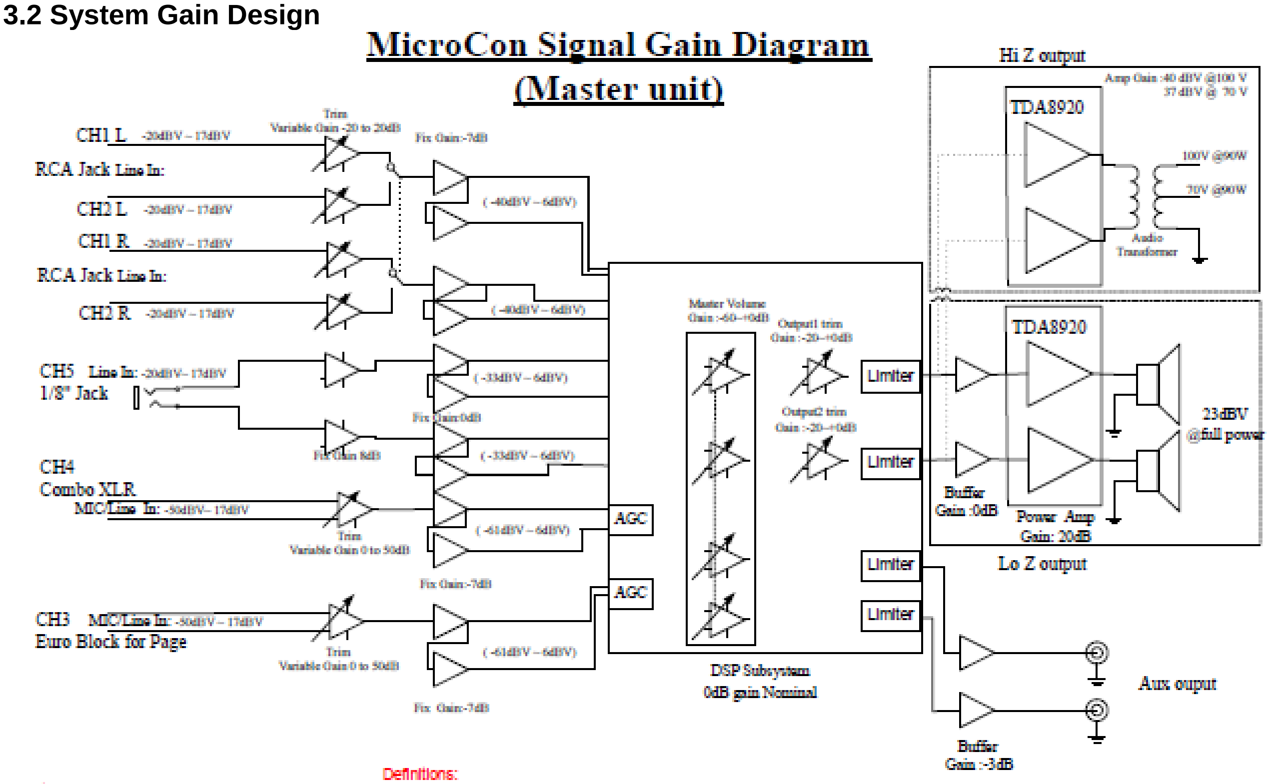


3. Detail System Features and performance

4.1 System Signal Flow Block Diagram



3.2 System Gain Design



Note:
Supply voltage of operational amplifiers: +/-12V

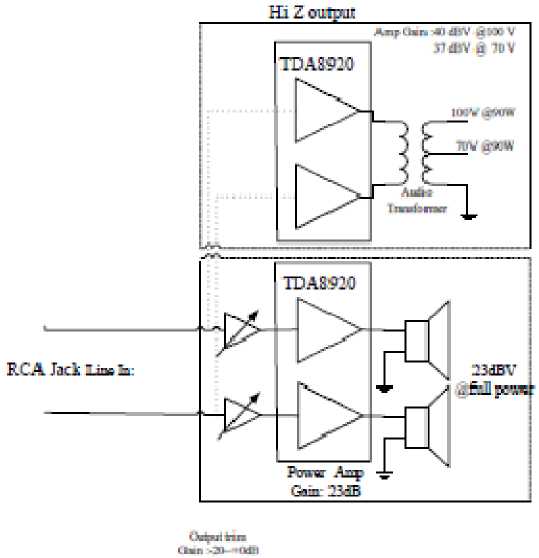
Definitions:
AGC
set point:+3dBV (can drive the amp to full power)

Clip level: - 3dB FS.

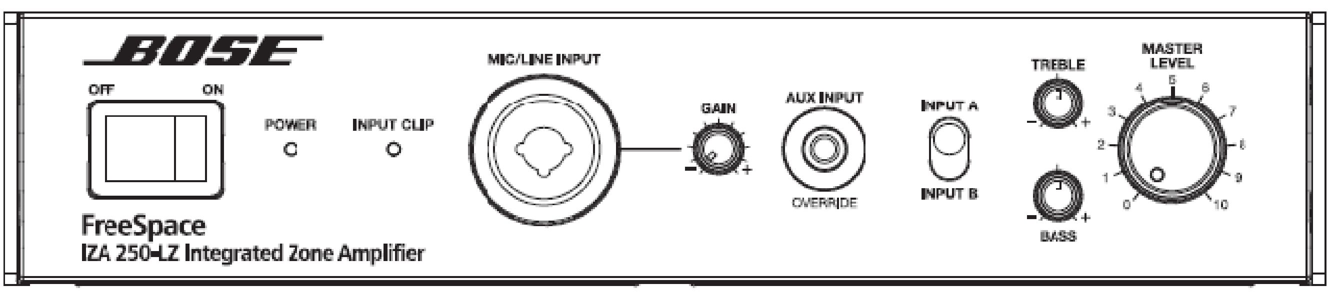
Limiter:
set point is 4dBV (Limit the amp at 60W – Lo Z unit)
3.5dBV (Limit the amp at 100W – Hi Z unit)

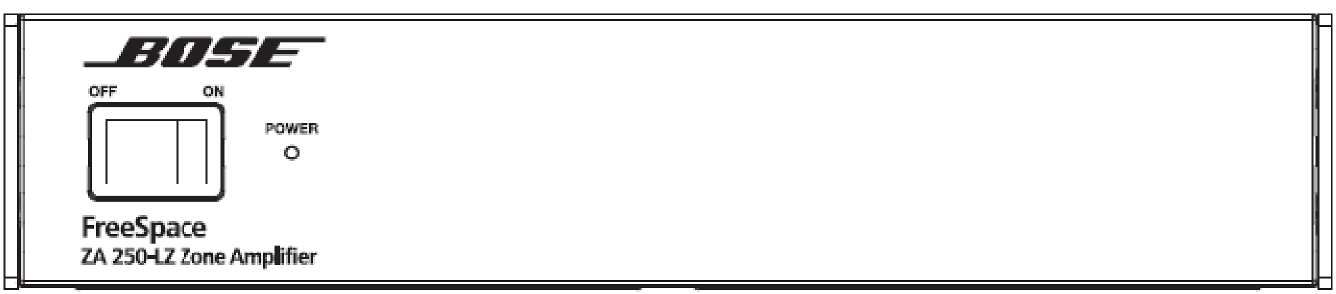
Ver: 0.40
Stage: MP1.4 or more higher stage
Author: Steven Xie/ J sun
Date: 2011-08-01

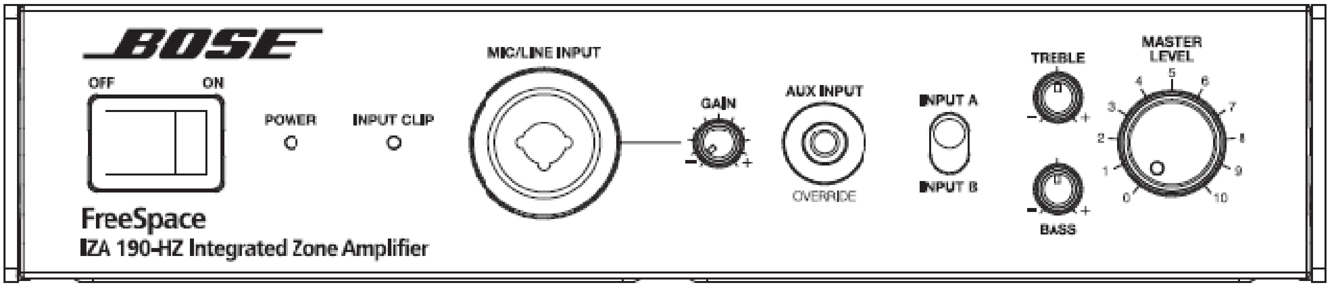
MicroCon Signal Gain Diagram
(Slave unit)

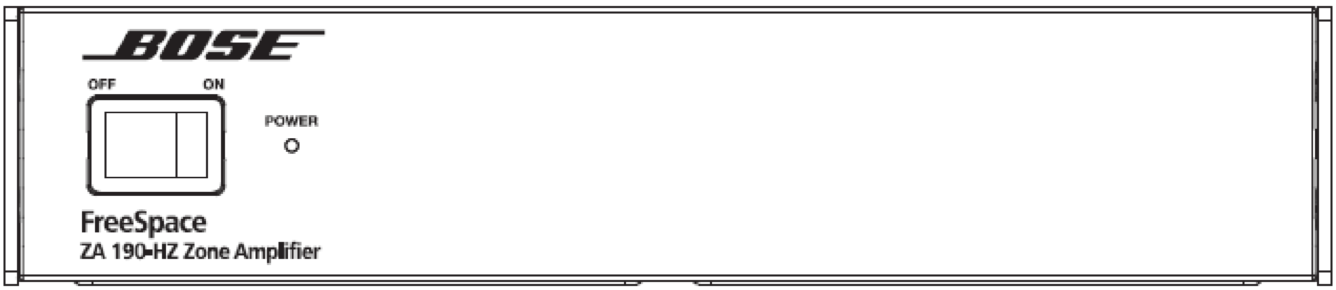


4. Front panel displays/control/connectors



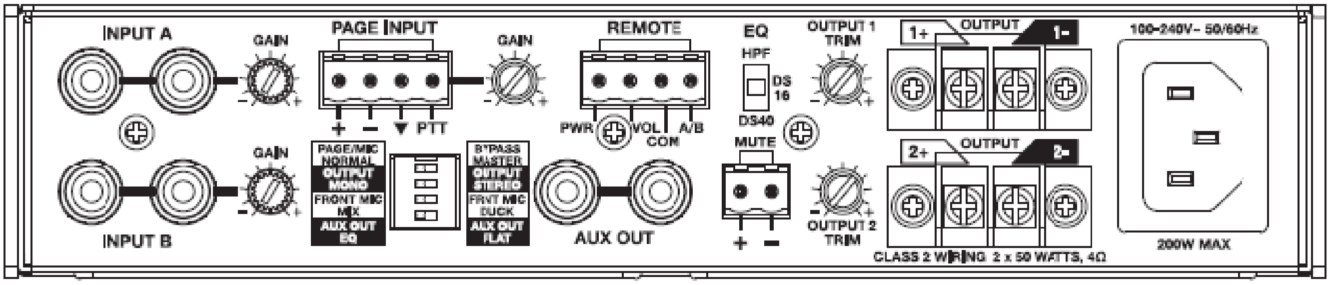


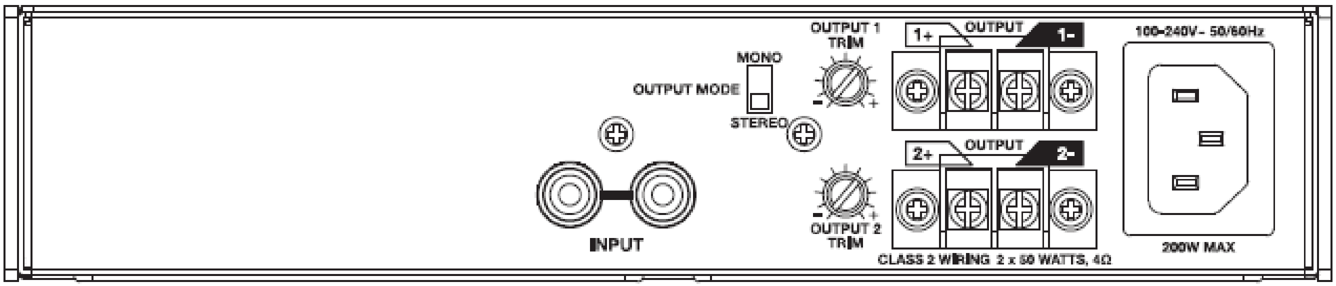


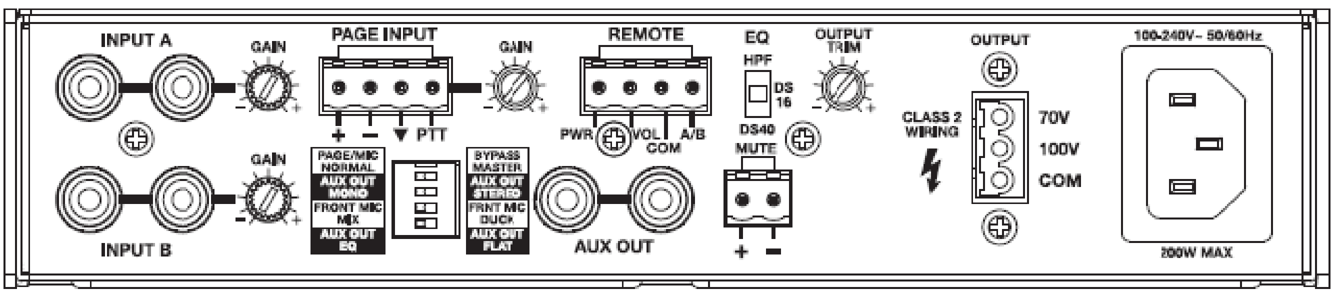


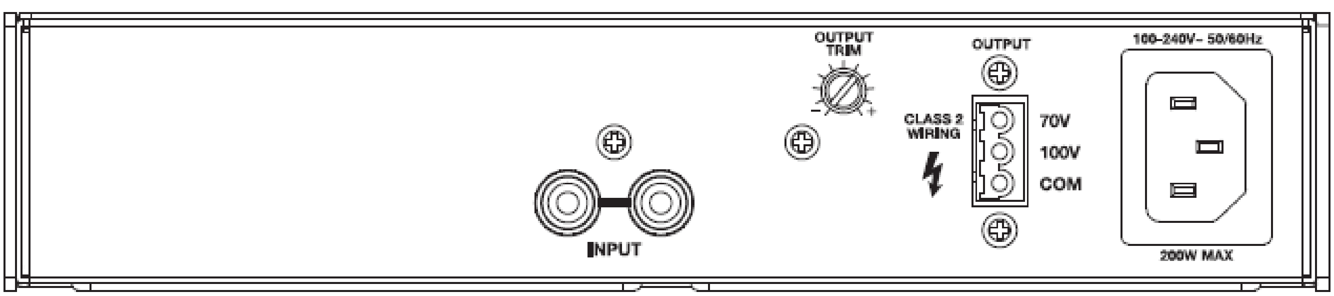
Item	Function	Lo Z		Hi Z	
		Main unit	Slave unit	Main unit	Slave unit
Power LED	Blue color LED	□	□	□	□
Power Switch	Rocker switch	□	□	□	□
Clip LED	Green color LED	□		□	
Mike/Line XLR Combo	XLR/Line Combo input Jack	□		□	
Combo in level	XLR/Line Combo Jack. Level control potentiometer	□		□	
Media Player Input	1/8” or 3.5mm mini phone jack	□		□	
Source Select	Slide switch two positions, select input 1 or input 2	□		□	
Output tone treble/bass	Control potentiometer	□		□	
Master Volume	Rotary Volume Potentiometer (use existing Bose Knob)	□		□	

5. Rear panel displays/control/connectors





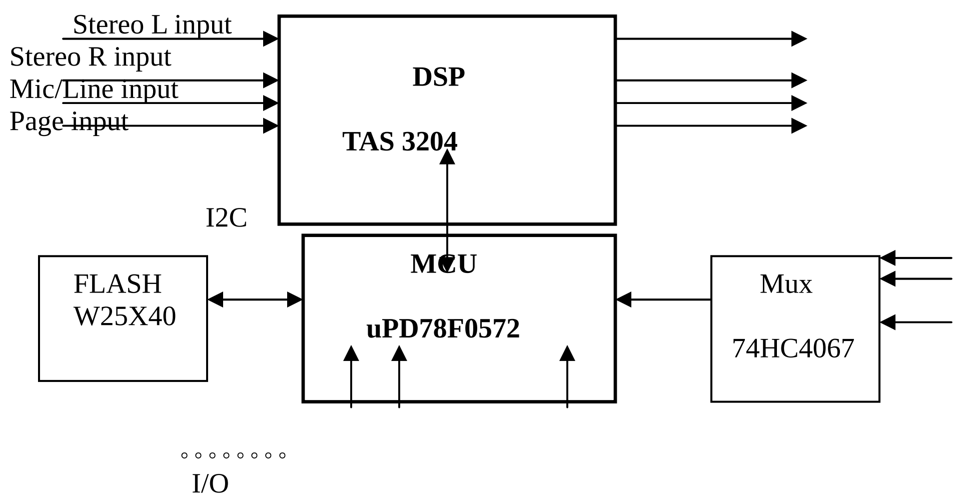




Item	Function	Lo Z		Hi Z	
		Main unit	Slave unit	Main unit	Slave unit
Inputs 1, 2	RCA jacks x 2 pairs	□		□	
RCA input level	Rotary gain control	□		□	
Input 3	4 pin green Euro Block for Paging input	□		□	
Page input level	Rotary Potentiometer	□		□	

Remote	4 pin black Euro Block for Remote control input	□		□	
Mute control	2 pin black Euro Block for Mute control input	∨		∨	
Aux output	RCA jacks x2	∨		∨	
Bypass On/Off switch Mon/Stereo switch MIC input MIX/DUCK switch AUX out EQ switch	4 position dip switch	∨		∨	
EQ selector	3 position slide switch (D16, D40, HPF)	∨		∨	
Output level	Rotary Potentiometer	∨	∨	∨	∨
Lo Z Output	3 pin Speaker terminal	∨		∨	
Hi Z Output	2 pin speaker terminal		∨		∨
AC Inlet	Snap-In ICE Connector	∨	∨	∨	∨
RCA input	RCA jacks x 2 pairs		∨		∨
Mono/Stereo switch	2 position slide switch		∨		

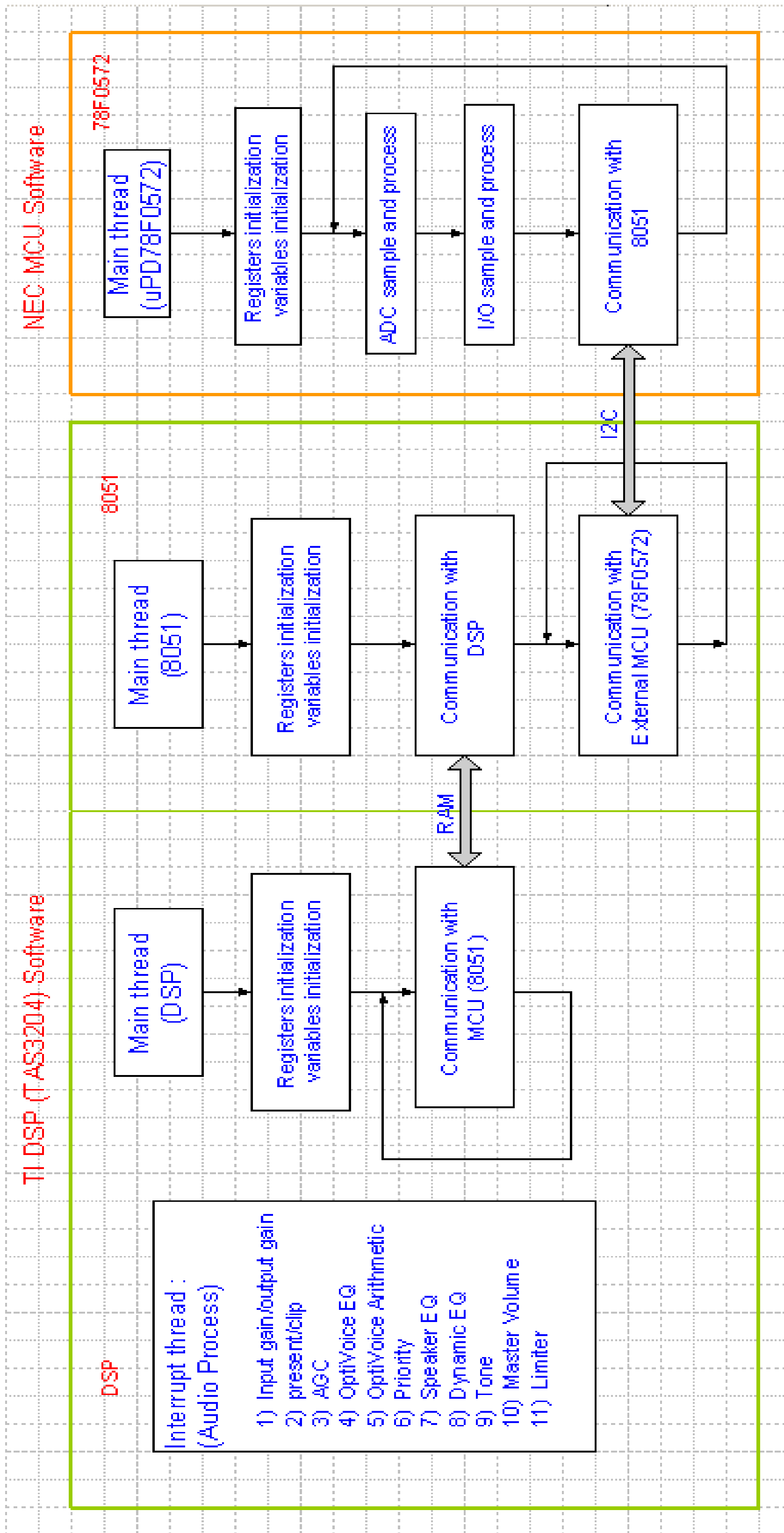
6. DSP Subsystem
6.1 DSP and MCU communication design



Hardware block diagram is shown as above Figure .

7.2 System Software Architecture

The system software consists of two parts: DSP and MCU, the DSP communicates with the MCU through an I2C bus. Refer to below system software architecture figure.



The TAS3204 has two cores in one chip, one is the DSP, the other is the 8051, so its software includes DSP's code and the 8051's code. In the DSP one core handles the main thread, the other is used for the interrupt thread that process audio effects. The 8051 handles communicating with main DSP and the external MCU. The external MCU samples the knob and switch positions on the panel and sends the values to 8051 through the I2C protocol.

DSP's interrupt processes below audio effects:

1) input/output gain
2) present/clip
3) AGC
4) Optivoice EQ
5) Optivoice arithmetic
6) Priority
7) Speaker EQ
8) Dynamic EQ
9) Tone
10) Master volume
11) Limiter

7.3 DSP internal signal flow and all modules requirement
7.3.1 Signal Flow Chart

This is a 4-ins, 4-outs system, see below Figure for the signal flow chart. Signal processing block will be described in the following sections.

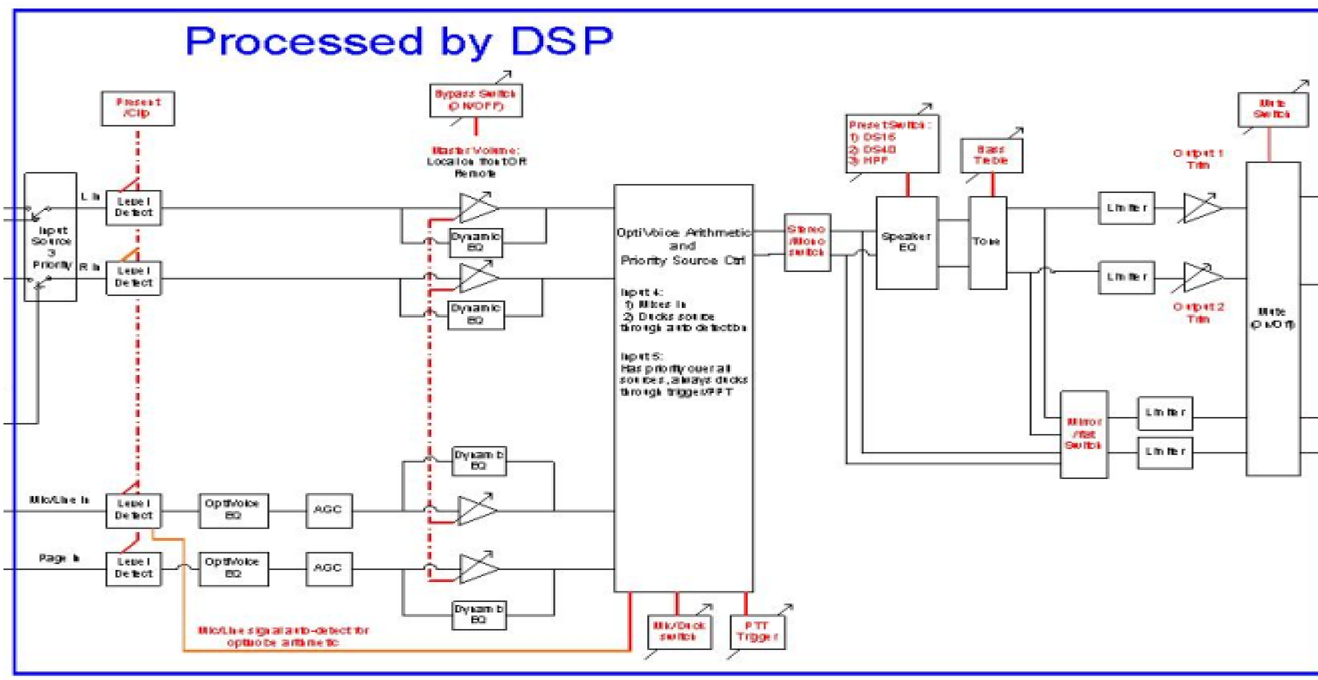


Figure: DSP Internal Signal Flow

7.3.2 The module requirement in DSP
7.3.2.1 Master Volume Control

Master Volume can be adjusted by knobs on the front panel.

Adjustment range is -∞ ~ +0dB, step is 1dB.

7.3.2.2 Signal Clip

All inputs share the one Clip LED on the front panel.

Signal clip: when the signal is too strong, and greater than another threshold, a red LED is on to indicate the signal is too large. The clip signal to DSP input is 3dBV.

Input clip signal source	Input 1/2		Input 3	Input 4		Input 5	
Input trim location	Min.	Max.	N/C	Min.	Max.	Min.	Max.
Clip Level (dBV)	> 10dBV	> -10dBV	> + 5dBV	> + 13dBV	> - 37dbV	> + 13dBv	> - 37dBV

Table. Input signal clip configuration

7.3.2.3 Auto Gain Control

Set point: default is -3dbfs (can drive the amp to full power)

gain: range is -10dB to +20dB.

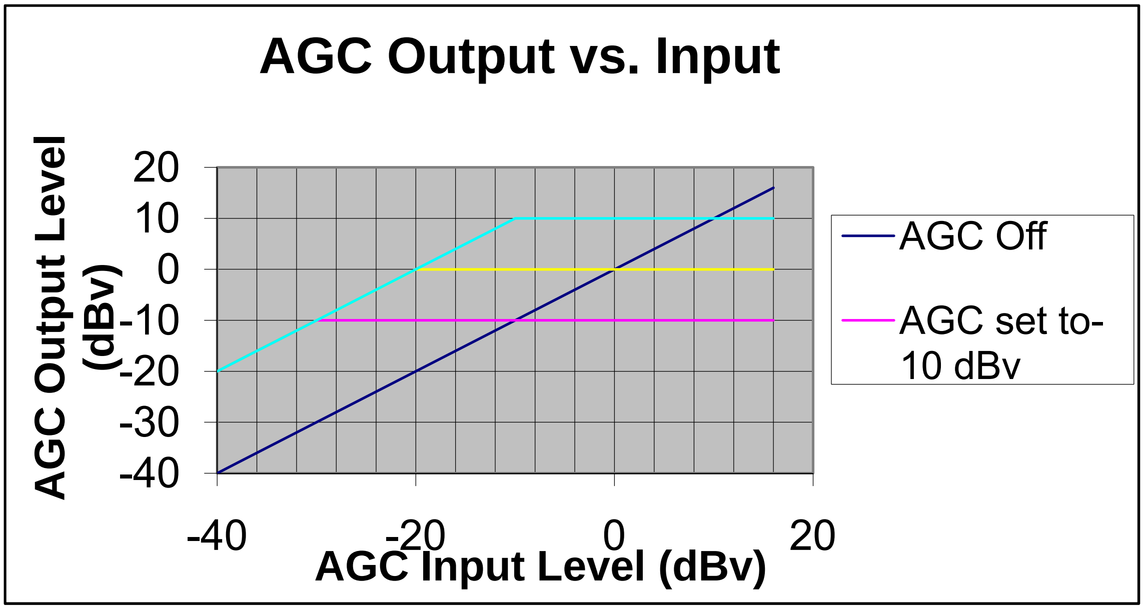
Attack time: 320ms

Release time: 500ms

Automatic Gain Control (AGC) Algorithm:

The AGC's primary function is to keep source material at a constant level. Consider two music passages recorded at different levels. With the AGC circuit enabled, the input gain is dynamically adjusted to ensure a constant playback level. Two average detectors are used for independent attack and decay settings. When no signal is present, the AGC will stay at its last gain setting.

The maximum AGC gain is limited to +20 dB to help ensure an 88 dB dynamic range. The AGC provides up to 30 dB of attenuation. When enabled, the AGC adjusts its gain (within the bounds –30 to +20 dB) to maintain a constant output level (known as the AGC set point). For the AGC, a feed-forward approach is used where the input signal is averaged subject to the attack and release parameters. A steady state level diagram for the AGC is seen below:



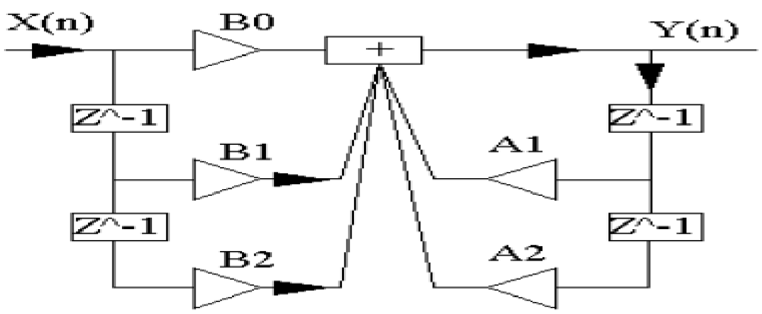
The gain of the AGC is simply the AGC set point (in dB) minus the current average value of the AGC input signal (in dB). Consider the steady state example where the input signal is –10 dB, and the AGC set point is +10 dB. The AGC gain is defined as: 10dB - (-10 dB) = 20 dB of gain

The ballistics of the gain change are defined by the AGC attack and release parameters as follows. Tau is either the attack or release time constant.

Gain(t) = (initial gain – final gain) exp(-t/Tau) + final gain*

7.3.2.4 Optivoice® EQ

Optivoice eq is a Band EQ(2 sequential 2nd IIR Butterworth filters) . 2nd IIR filter's architecture, please refer to below figure.



Fc = 1.6KHz, Q=1, gain=3dB

7.3.2.5 Opti-Voice Arithmetic

The Opti-Voice algorithm only works in the PTT trigger mode.

In PTT (Push to talk) mode, the trigger condition is exactly the PTT contact closure input signal. If this PTT signal is not asserted, signal(usually music) from the input is not attenuated, page input (usually speech) is attenuated to NEG unlimited dB. If the PTT signal is asserted, the music goes down and speech passes. When the PTT signal ends, speech gets muted immediately and the music signal waits about 1 second and then ramps up.

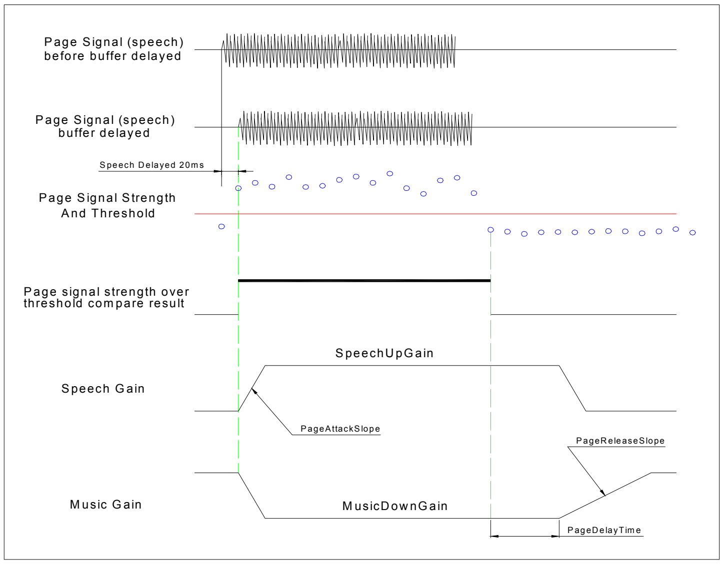


Figure : OPTI-Voice Paging

System default setting:

speech delay 2ms
attack time 1ms
release time 2s
duck depth -20dB
mic auto level detect threshold: -44dB

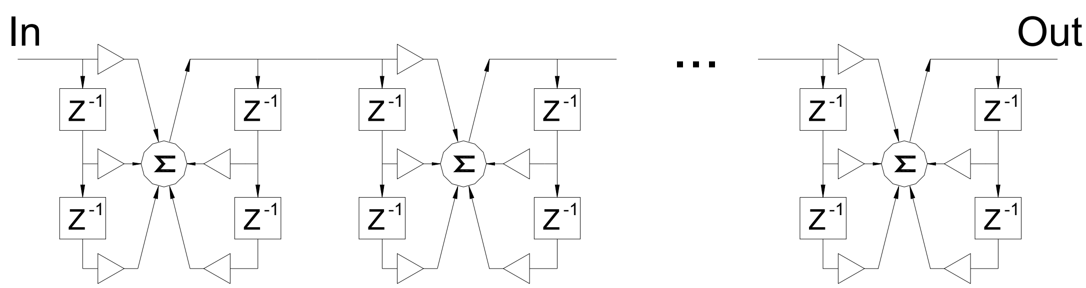
7.3.2.6 Priority

When Input 5 (page input) is activated, it takes priority over all inputs and ducks the other sources.
When Input 4(mic/line input) is activated, the front mic/line input can either be mixed without ducking or override the sources and duck the sources. Requires auto detect for ducking.
Switch for on/off option of priority mic/line input and page input, being subject to master volume or not.

7.3.2.7 Bose® Speaker EQ

The Bose Speaker EQ contains 13 bi-quad sections(13 sequential 2nd IIR filters). So each BOSE Speaker EQ block has 65 IIR filter coefficients. There are 3 sets of pre-stored EQ coefficients setting in the system, which one to be activated is controlled by either preset select switch.

Figure 4: BOSE® Speaker EQ



Each 2nd IIR filters has 5 coefficients (b0,b1,b2,a1,a2), and every coefficient occupy 4 bytes. So 1 set of coefficient will need memory:

13 * 5 * 4 = 260bytes

In this system, there are 3 sets of presets for three different speaker, they are:

1) DS16
2) DS40
3) HPF (40Hz for Low-Z, 60Hz for Hi-Z)

7.3.2.8 Output Volume Control And Dynamic EQ

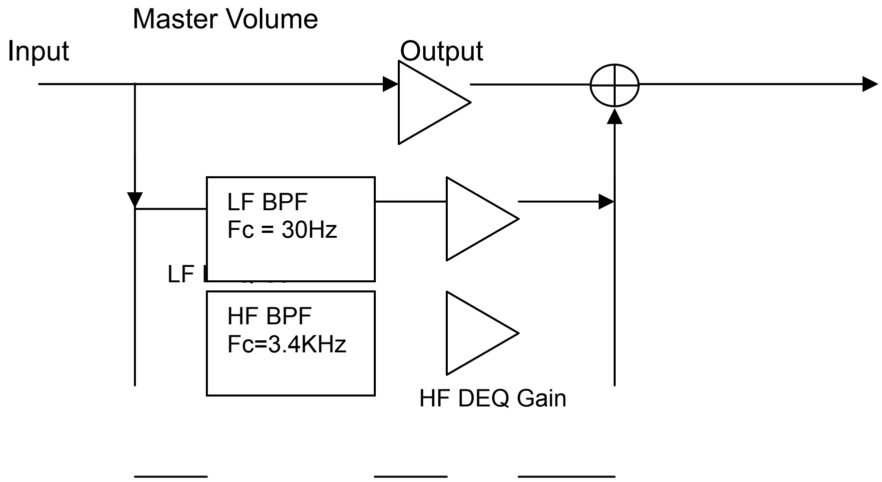


Figure : Dynamic EQ and Gain Control

The LF BFP and HF BPF gain index is automatically selected by the DSP according to the output Gain value. Refer to the chart below.

Index	Gain Setting	LF DEQ Gain	HF DEQ Gain
0	0 to -3 dB	0.000000000000	0.000000000000
1	-3 to -6 dB	0.058784476731	0.028452529582
2	-6 to -9 dB	0.086688059682	0.041095243184
3	-9 to -12 dB	0.095928387908	0.044522512384
4	-12 to -15 dB	0.098717433667	0.041784934728
5	-15 to -18 dB	0.095488458680	0.036711532079
6	-18 to -21 dB	0.087598619122	0.031210996428
7	-21 to -24 dB	0.079164344399	0.025690480373
8	-24 to -27 dB	0.072005506223	0.020481463209
9	-27 to -30 dB	0.063789696291	0.016169623939
10	-30 to -33 dB	0.055446370549	0.012662749810
11	-33 to -36 dB	0.047528856310	0.009643378488
12	-36 to -39 dB	0.040293303092	0.007317975432
13	-39 to -42 dB	0.033861735201	0.005535847694
14	-42 to -45 dB	0.028516201002	0.004121366456
15	-45 to -48 dB	0.023968041852	0.003043777748
16	-48 to -51 dB	0.020036114901	0.002245400449
17	-51 to -54dB	0.016734711819	0.001643948239
18	-54 to -57 dB	0.014021841837	0.001187422820
19	-57 to -60 dB	0.011708026166	0.000857460868
20	-60 to -63 dB	0.009747834967	0.000619039450

Table : Dynamic EQ Coefficients

	LF BPF	HF BPF
b0	0.00214753846501	0.17338562315138
b1	-0.00002993588615	-0.00241693099323
b2	-0.00211760257885	-0.17096869215815
a0	1.00000000000000	1.00000000000000
a1	-1.99571662856685	-1.47215238262537
a2	0.99573485895566	0.65564518709885

7.3.2.9 Output Tone Control (Bass and Treble)

Tone Control is a 2 Band EQ (2 sequential 2^{nd} IIR Butterworth filters), whose band centers are 100Hz (Bass) and 7KHz (Treble). Bass range is –6dB~+6dB, Treble range is –6dB~+6dB, and the adjust step for both Bass and Treble is 0.25dB, i.e. total 49 steps. Out1, Out2 each has a Tone Control.

Table: Output Tone Control Configuration

	Out1	Out2
Tone Control Needed or not	Needed	Needed
Fc	135Hz, 5.5KHz	135Hz, 5.5KHz
Bass gain range	-6dB ~ +6dB	-6dB ~ +6dB
Treble gian range	-6dB ~ +6dB	-6dB ~ +6dB
Bass step	0.25dB, 49 steps	0.25dB, 49 steps
Treble step	0.25dB, 49 steps	0.25dB, 49 steps
Frequency Response symmetry	symmetry needed	symmetry needed
Control	Knobs on front panel	Knobs on front panel

7.3.2.10 Limiter

Default setting: Setpoint : 3.7dBV for Hi-Z, 4dBV for Low-Z
Attacktime :2ms
Releasetime: 5s

7.4 MCU Software

MCU is responsible for the interface processes, it samples the position of the switches and knobs, then translates to a digital value and then sends these values to the DSP.

7.4.1 I/O control

The I/O signals are listed in the following table.

	Input/output Flags	**Functions**
1	INPUT1/INPUT2 Select switch	Select which one input1 or input2
2	PTT	Push to talk
3	EQ -- HPF	Speaker EQ
4	EQ – DS16	Speaker EQ
5	EQ – DS40	Speaker EQ
6	MIX/DUCK switch	Select mix or duck mode
7	Mono/Stereo switch	Select Mono or stereo
8	Bypass master volume/ no bypass switch	Select bypass or no bypass
9	Mute detect	Mute detect
10	Present/ clip	Indicate signal's present or clip
11	Remote_input1/2 position	Detect input1/2 switch position on the remote
12	Control 74HC4053 for select input1/2	for select input1/2

Table. I/O Control

7.4.2 ADC Sample

	ADC Sample	**Functions**
1	Remote master volume	Sample master volume on the remote
2	Output1 gain	Adjust output1 gain
3	Output2 gain	Adjust output2 gain
4	treble	Adjust output tone gain
5	bass	Adjust output tone gain
6	Master level	Adjust master level gain

Table. ADC Sample

7.7.3 Communication with DSP

The MCU uses the I2C protocol to communicate with the DSP. After the MCU samples the value of the switches and knobs, it sends them to DSP.

7.4.4 Boot DSP source code

The DSP source code is placed into FLASH memory, and this flash is mounted on the MCU board, so when the power is applied, the MCU must read the DSP's source code from the flash memory, then send it to the DSP. After the DSP receives the code correctly, the DSP starts to run the program.

8. Analog Inputs, Output, and User interface Subsystem

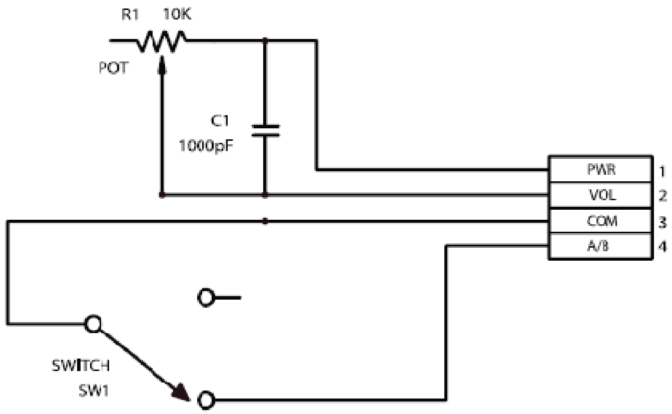
8.1 Line Inputs

Specification	**Nominal**	**Limits**	**Notes**
input impedance, differential	> 20K ohms	+/-10%	at 1kHz
CMRR referred to output	70db	>=60dB	at 1kHz, 20dB gain, 200 ohm source impedance
Input sensitivity for codec 0dBFS	-20dBV to +17dBV	+/-1dB	At 1kHz
Gain Range	20B to –20dB	+/-1dB	1kHz, 50 ohm source
maximum input level	+17dBV	>=+17dBV	THD+N <=0.3%, 60-16kHz, 0dB gain
Output Noise		-65dBV	Measured at amp output, Line input gain at +20 dB

8.2 Mic Inputs:

Specification	**Nominal**	**Limits**	**Notes**
input impedance, differential	2K ohms	+/-10%	at 1kHz
Input sensitivity for codec FS	-50dBV to +17dBV	+/-1dB	At 1kHz
gain, Range	0 dB to 50dB	+/-1dB	1kHz, 200 ohm source
maximum input level	+17dBV	>=10dBV	THD+N <=0.3%, 60-16kHz, 0dB gain
Output Noise		-65dBv	Measured at amp output, Line input gain at +50 dB

8.3 Wall Plate Control Interface



Detailed schematic for volume control with A/B select user interface

8.4 Behavior of Mono Stereo Switch

Assume the input is plugged into line 1 and is a stereo source. The table below shows what is expected at the line and amplifier outputs for Mono and Stereo modes.

	Low Z - Stereo mode Line Output L	Line Output R	Amp Out 1	Amp Out 2
Line Input - 1L	1L		1L	
Line Input - 1R		1R		1R

	Low Z - Mono mode Line Output L	Line Output R	Amp Out 1	Amp Out 2
Line Input - 1L	1L+1R		1L+1R	
Line Input - 1R		1L+1R		1L+1R

	High Z - Stereo mode Line Output L	Line Output R	Amp Out 1
Line Input - 1L	1L		1L
Line Input - 1R		1R	

	High Z - Mono mode Line Output L	Line Output R	Amp Out 1
Line Input - 1L	1L+1R		1L+1R
Line Input - 1R		1L+1R	