National Semiconductor

February 26, 2008

LMV1088
Dual Input, Far Field Noise Suppression Microphone Amplifier with Automatic Calibration Ability

General Description

The LMV1088 is a fully analog dual input microphone array amplifier designed to reduce background acoustic noise, while delivering superb speech clarity in voice communications applications.

Part of the PowerWise® family of energy efficient solutions, the LMV1088 incorporates calibration circuitry which may be initiated by either an I²C command or by a logic level control on a separate input pin. The calibration sequence compensates for gain and frequency response variations of the microphones used with the LMV1088, eliminating the need to use expensive matched microphone sets. The calibration data is stored in the internal EEPROM memory. The LMV1088 has two differential input microphone amplifier channels plus far field noise suppression (FFNS) processing circuitry. The amplifiers and FFNS circuitry are adjustable for gain differences in the MIC channels of +/- 3dB. The frequency response variations of the microphones over the voice band frequency range can also be adjusted for differences of +/-3dB.

The compensation or calibration function is achieved via memory stored coefficients. These are determined when the FFNS calibration fuction is activated. The purpose of the calibration sequence is to choose the optimized coefficients for the FFNS circuitry for the given microphones, spacing, and acoustical environment.

Key Specifications

(3.3V supply, unless otherwise specified)

- Supply voltage 2.7V to 5.5V
- Supply current 1mA (typ)
- Signal to noise ratio (A-weighted) 60dB (typ)
- Total harmonic distortion (A-weighted) 0.1% (typ)
- Temperature range –40°C to 85°C

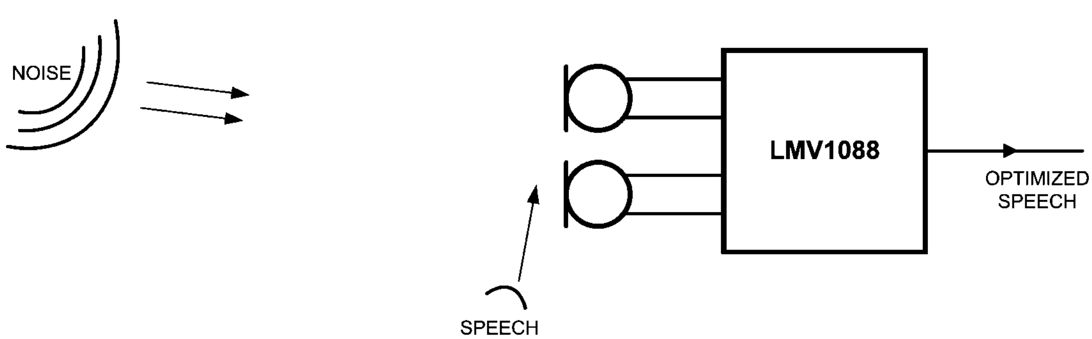
Features

- Low power consumption
- Neglectable noise suppression processing delay
- Automatic Calibration
- Three microphone usage modes
- Space-saving 36 Bump micro SMD package

Applications

- Cellular phones
- Mobile and handheld two-way radios
- Bluetooth and other powered headsets

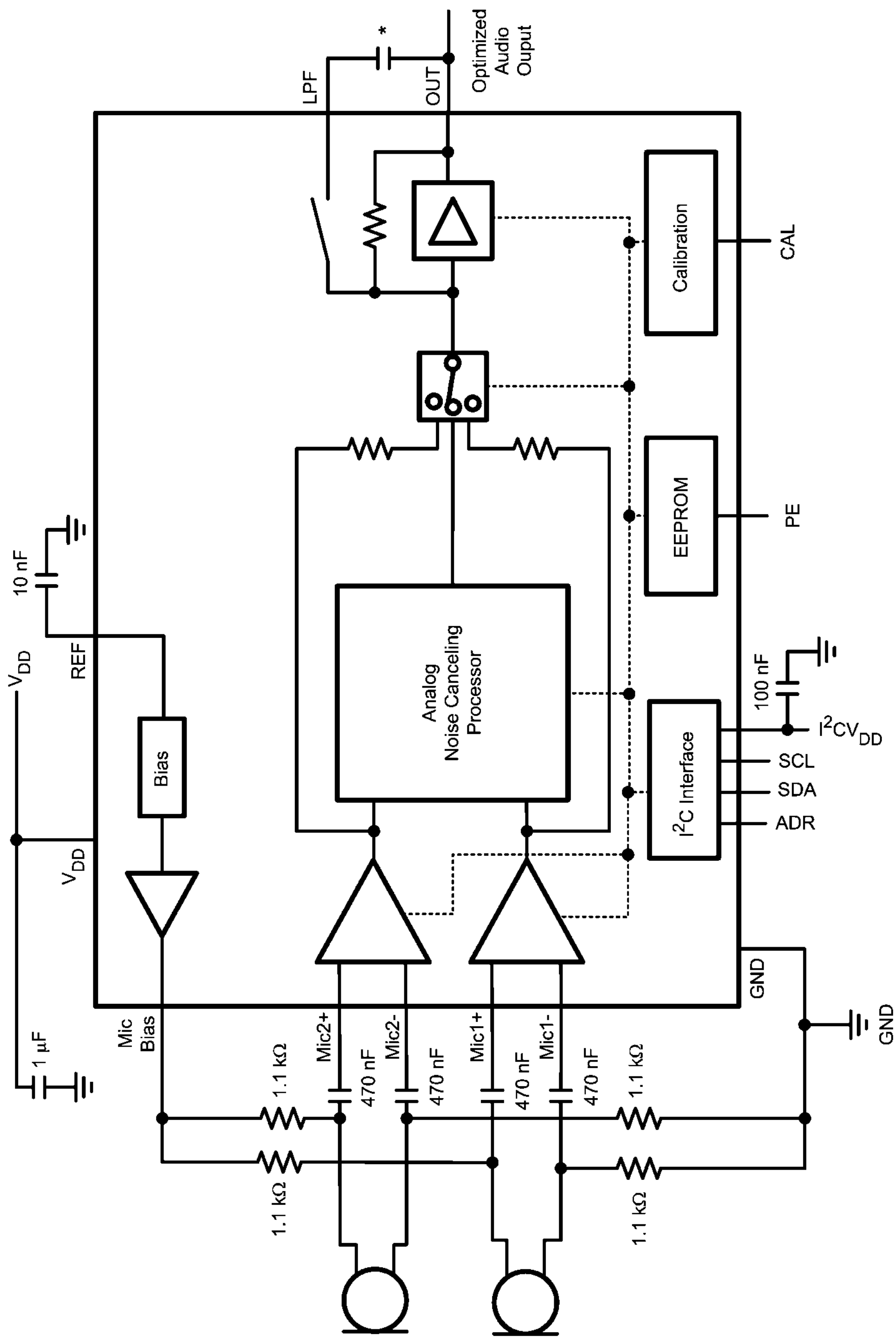
Application of the LMV1088



20213040

PowerWise® is a registered trademark of National Semiconductor.

Typical Application

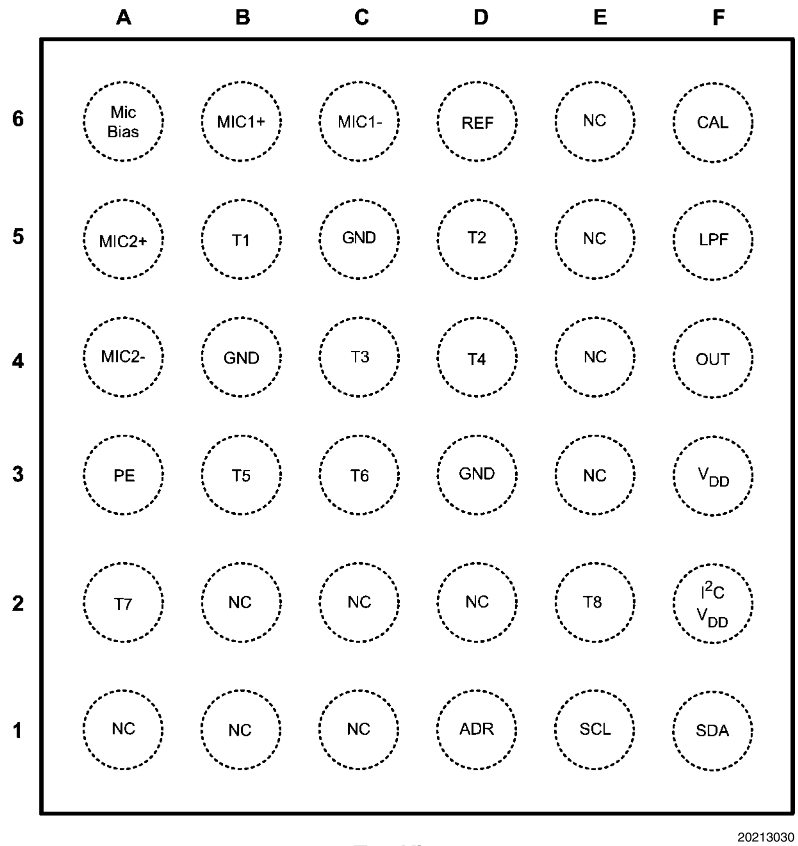


* The value of the low pass filter capacitor is application dependent, see the application section for additional information.

FIGURE 1. Typical Dual Microphone Far Field noise Cancelling Application

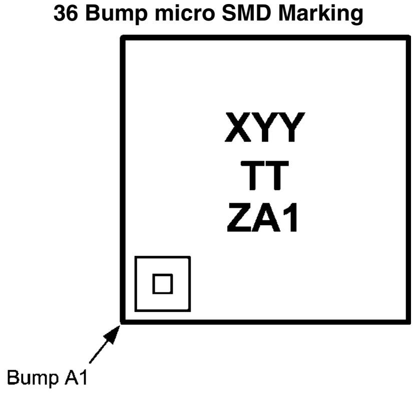
Connection Diagrams

36 Bump micro SMD package



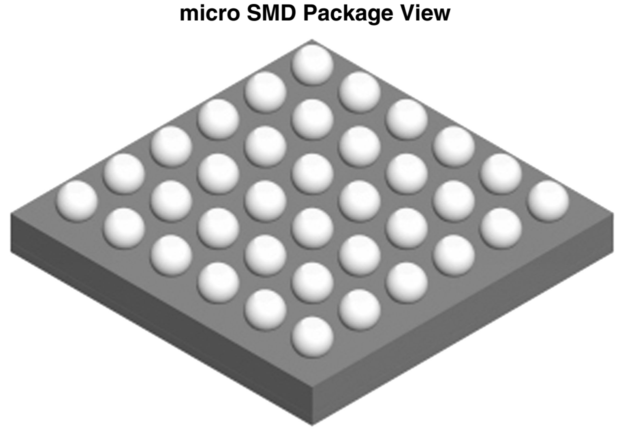
Top View
Order Number LMV1088RL
See NS Package Number RLA36VVA

36 Bump micro SMD Marking



Top View
X = Plant Code
YY = Date Code
TT= Die Tracability
ZA1 = LMV1088RL

micro SMD Package View



Bottom View

TABLE 1. Pin Name and Function

Bump Number	Pin Name	Pin Function
A1	NC	No Connect (Note 1)
A2	T7	Connect to GND
A3	PE	Program Enable EEPROM
A4	MIC2–	microphone 2 input —
A5	MIC2+	microphone 2 input +
A6	Mic Bias	Bias for Microphones
B1	NC	No Connect (Note 1)
B2	NC	No Connect (Note 1)
B3	T5	Float(Note 2)
B4	GND	Amplifier ground
B5	T1	Float(Note 2)
B6	MIC1+	microphone 1 input +
C1	NC	No Connect (Note 1)
C2	NC	No Connect (Note 1)
C3	T6	Float(Note 2)
C4	T3	Float(Note 2)
C5	GND	Amplifier ground
C6	MIC1–	microphone 1 input —
D1	ADR	I^2C Address select
D2	NC	No Connect (Note 1)
D3	GND	Amplifier ground
D4	T4	Float(Note 2)
D5	T2	Connect to GND
D6	REF	Reference Voltage De-coupling
E1	SCL	I^2C Clock
E2	T8	Connect to GND
E3	NC	No Connect (Note 1)
E4	NC	No Connect (Note 1)
E5	NC	No Connect (Note 1)
E6	NC	No Connect (Note 1)
F1	SDA	I^2C Data
F2	I^2CV_{DD}	I^2C power supply
F3	V_{DD}	Power Supply
F4	OUT	Optimized Audio Out
F5	LPF	Lowpasss Filter Capacitor
F6	CAL	Calibration Start

Note 1: Connect NC pins to GND for optimum noise performance

Note 2: Grounding the float pins can result in excessive currents

Absolute Maximum Ratings (Note 3)

If Military/Aerospace specified devices are required, please contact the National Semiconductor Sales Office/ Distributors for availability and specifications.

Supply Voltage	6.0V
Storage Temperature	-85°C to +150°C
ESD Rating (Note 6)	2000V
ESD Rating (Note 7)	200V
Junction Temperature (T_{JMAX})	150°C
Mounting Temperature Infrared or Convection (20 sec.)	235°C
Thermal Resistance	
θ_{JA} (microSMD)	70°C/W

Soldering Information See AN-112 "microSMD Wafers Level Chip Scale Package."

Operating Ratings (Note 4)

Supply Voltage	2.7V to 5.5V
I^2CV_{DD} (Note 12)	1.8V to 5.5V
Temperature Range	–40°C to 85°C

Electrical Characteristics 3.3V (Note 3)

Unless otherwise specified, all limits guaranteed for $T_J = 25°C$, $V_{DD} = 3.3V$, $V_{IN} = 18mV_{PP}$, pass through mode (Note 10), preamplifier gain = 20 dB, postamplifier gain = -2.5dB, $R_L = 100k\Omega$, and $C_L = 4.7pF$.

Symbol	Parameter	Conditions	LMV1088		Units (Limits)
			Typical (Note 8)	Limits (Note 9)	
SNR	Signal-to-Noise Ratio	f = 1kHz, , $V_{IN} = 18mV_{PP}$, A-Weighted	60		dB
V_{IN}	Max Input Signal	f = 1kHz and THD+N < 1%	97		mV_{PP}
V_{out}	AC Output Voltage	f = 1kHz	500		mV_{RMS}
	DC Output Voltage		800		mV
THD+N	Total Harmonic Distortion + Noise	f = 1kHz, $V_{IN} = 18mV_{PP}$	0.1		%
Z_{IN}	Input Impedance		100		kΩ
Z_{OUT}	Output Impedance		150		Ω
Z_{LOAD}		R_{LOAD} C_{LOAD}		10 10	kΩ (min) pF (max)
A_M	Microphone Pre Amplifier Gain Range	f = 1kHz	6 – 36		dB
A_{MR}	Microphone Pre Amplifier Gain Adjustment Resolution	f = 1kHz	2		dB
A_P	Post Amplifier Gain Range	f = 1kHz Pass Through Mode and Summing Mode	-2.5 – 9.5		dB
		f = 1kHz Noise Cancelling Mode (Note 11)	0 – 12		dB
A_{PR}	Post Amplifier Gain Adjustment Resolution	f = 1kHz	3		dB
A_{CR}	Gain Compensation Range	f = 300Hz — f = 3400Hz		±3	dB (max)
A_{MD}	Gain Matching Difference After Calibration	f = 300Hz f = 1kHz f = 3kHz		0.5 0.5 0.5	dB (max) dB (max) dB (max)
T_{CAL}	Calibration Duration			770	ms (max)
PSRR	Power Supply Rejection Ratio	Input Referred, Input AC grounded			
		f = 217Hz (100mV_{PP})	85		dB
		f = 1kHz (100mV_{PP})	80		dB
CMRR	Common Mode Rejection Ratio	f = 1kHz,	60		dB
V_{BM}	Microphone Bias Supply Voltage	$I_{BIAS} = 1mA$	2.0		V
ε_{VBM}	Microphone Bias Supply Noise	A-Weighted	10		μV_{RMS}
I_{BM}	Total available Microphone Bias Current			1.2	mA (min)
I_{DDQ}	Supply Quiescence Current	$V_{IN} = 0V$	1	1.5	mA (max)
I_{DDCP}	Supply Current during Calibration and Programming	Calibrating or Programming EEPROM	28	50	mA (max)
I_{DD}	Supply Current	$V_{in} = 25mV_{PP}$ both inputs, Noise canceling mode	1	1.5	mA (max)

Electrical Characteristics 5.0V (Note 3)

Unless otherwise specified, all limits guaranteed for $T_J = 25°C$, $V_{DD} = 5V$, $V_{IN} = 18mV_{PP}$, pass through mode (Note 10), preamplifier gain = 20dB, postamplifier gain = –2.5dB, $R_L = 100kΩ$, and $C_L = 4.7pF$.

Symbol	Parameter	Conditions	LMV1088		Units (Limits)
			Typical (Note 8)	Limit (Note 9)	
SNR	Signal-to-Noise Ratio	f = 1kHz,, V_{IN} = 18mV_{PP}, A-Weighted	60		dB
V_{IN}	Max Input Signal	f = 1kHz and THD+N < 1%	97		mV_{PP}
V_{out}	AC Output Voltage	f = 1kHz	500		mV_{RMS}
	DC Output Voltage		800		mV
THD+N	Total Harmonic Distortion + Noise	f = 1kHz V_{IN} = 18mV_{PP}	0.1		%
Z_{IN}	Input Impedance		100		kΩ
Z_{OUT}	Output Impedance		150		Ω
Z_{LOAD}		R_{LOAD} C_{LOAD}		10 10	kΩ (min) pF (max)
A_M	Microphone Pre Amplifier Gain Range	f = 1kHz	6 – 36		dB
A_{MR}	Microphone Pre Amplifier Gain Adjustment Resolution	f = 1kHz	2		dB
A_P	Post Amplifier Gain Range	f = 1kHz Pass Through Mode and Summing Mode	-2.5 – 9.5		dB
		f = 1kHz Noise Cancelling Mode (Note 11)	0 – 12		dB
A_{PR}	Post Amplifier Gain Adjustment Resolution	f = 1kHz	3		dB
A_{CR}	Gain Compensation Range	f = 300Hz — f = 3400Hz		±3	dB (max)
A_{MD}	Gain Matching Difference After Calibration	f = 300Hz f = 1kHz f = 3kHz		0.5 0.5 0.5	dB (max) dB (max) dB (max)
T_{CAL}	Calibration Duration			770	ms
PSRR	Power Supply Rejection Ratio	Input Referred, Input AC grounded			
		f = 217Hz (100mV_{PP})	85		dB
		f = 1kHz (100mV_{PP})	80		dB
CMRR	Common Mode Rejection Ratio	f = 1kHz	60		dB
V_{BM}	Microphone Bias Supply Voltage	I_{BIAS} = 1mA	2.0		V
$ε_{VBM}$	Microphone Bias Supply Noise	A-Weighted	10		μV_{RMS}
I_{BM}	Total Available Microphone Bias Current			1.2	mA (min)
I_{DDQ}	Supply Quiescence Current	V_{IN} = 0V	1	1.5	mA (max)
I_{DDCP}	Supply Current during Calibration and Programming	Calibrating or Programming EEPROM	28	50	mA(max)
I_{DD}	Supply Current	V_{in} = 25mV_{PP} both inputs, Noise canceling mode	1	1.5	mA (max)

Digital Interface Characteristics (Notes 3, 12)

Unless otherwise specified, all limits guaranteed for T_J = 25°C, I^2CV_{DD} within the Operating Rating (Note 12)

Symbol	Parameter	Conditions	LMV1088		Units (Limits)
			Typical (Note 8)	Limits (Note 9)	
V_{IH}	Logic High Input Level	SCL, SDA, ADR, CAL, PE pins		$0.6xI^2CV_{DD}$	V (min)
V_{IL}	Logic Low Input Level	SCL, SDA, ADR, CAL, PE pins		$0.4xI^2CV_{DD}$	V (max)
ts_{CAL}	CAL Setup Time		2		ms
th_{CAL}	CAL Hold time until calibration is finished			770	ms (min)
ts_{PEC}	PE Setup Time		2		ms
th_{PEC}	PE Hold until calibration is finished			770	ms (min)

Note 3: "*Absolute Maximum Ratings*" indicate limits beyond which damage to the device may occur, including inoperability and degradation of device reliability and/or performance. Functional operation of the device and/or non-degradation at the *Absolute Maximum* Ratings or other conditions beyond those indicated in the *Recommended Operating Conditions* is not implied. The *Recommended Operating Conditions* indicate conditions at which the device is functional and the device should not be operated beyond such conditions. All voltages are measured with respect to the ground pin, unless otherwise specified.

Note 4: *The Electrical Characteristics* tables list guaranteed specifications under the listed *Recommended Operating Conditions* except as otherwise modified or specified by the *Electrical Characteristics Conditions* and/or Notes. Typical specifications are estimations only and are not guaranteed.

Note 5: The maximum power dissipation must be de-rated at elevated temperatures and is dictated by TJMAX, θJC, and the ambient temperature TA. The maximum allowable power dissipation is PDMAX = (TJMAX –TA)/ θJA or the number given in the *Absolute Maximum Ratings*, whichever is lower. For the LMV1088, TJMAX = 150°C and the typical θJA for this microSMD package is 70°C/W and for the LLP package θJA is 64°C/W Refer to the Thermal Considerations section for more information.

Note 6: Human body model, applicable std. JESD22-A114C.

Note 7: Machine model, applicable std. JESD22-A115-A.

Note 8: Typical values represent most likely parametric norms at TA = +25°C, and at the Recommended Operation Conditions at the time of product characterization and are not guaranteed.

Note 9: Datasheet min/max specification limits are guaranteed by test, or statistical analysis.

Note 10: In Pass Though mode, only one microphone input is active. See also *I²C Compatible Interface* for more information how to configure the LMV1088

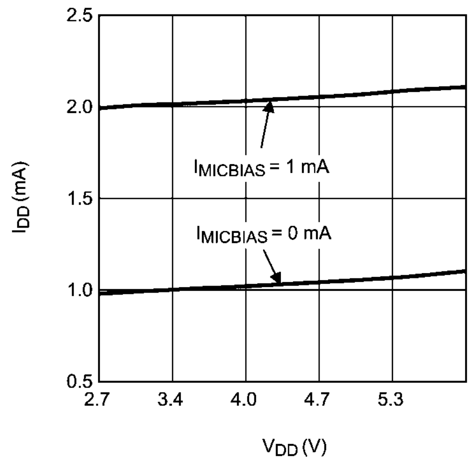
Note 11: In Noise Cancelling Mode there is 2.5 dB additional gain before calibration when compared to the other operating modes to compensate for the gain reduction that is caused by the noise cancelling effect

Note 12: The voltage at I^2CV_{DD} must not exceed the voltage on V_{DD}

Typical Performance Characteristics

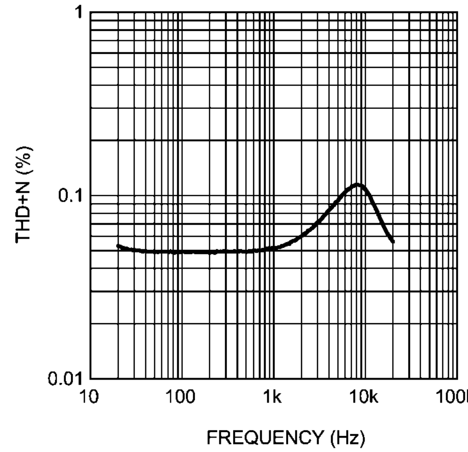
Unless otherwise specified, T_J = 25°C, V_{DD} = 3.3V, V_{IN} = 18mV_{PP}, pass through mode (Note 10), preamplifier gain = 20dB, postamplifier gain = –2.5dB, R_L = 100kΩ, and C_L = 4.7pF.

Supply Current vs. Supply Voltage



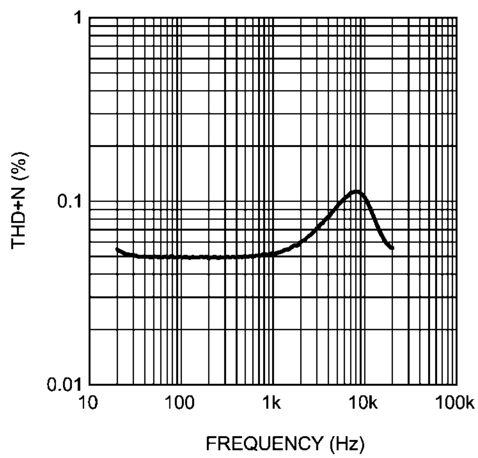
20213021

THD+N vs Frequency, pass trough mode Mic1 V_{IN} = 36mV_{pp}



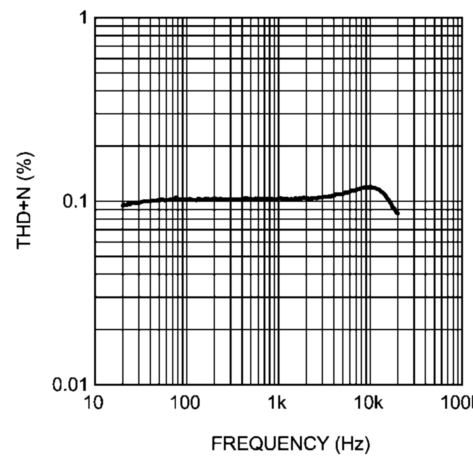
20213003

THD+N vs Frequency, pass trough mode Mic2 V_{IN} = 36mV_{pp}



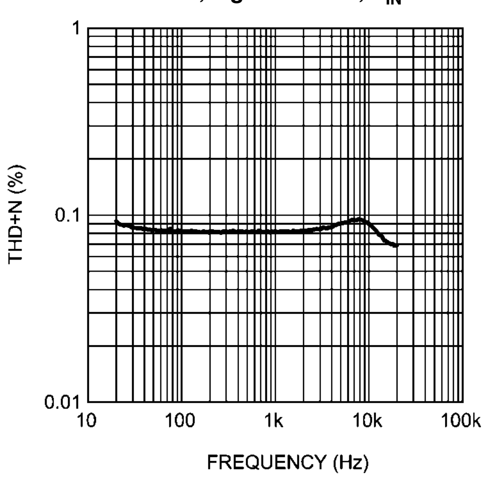
20213004

THD+N vs Frequency, Noise canceling mode signal at Mic1, Mic2 AC shorted, V_{IN} = 36mV_{pp}



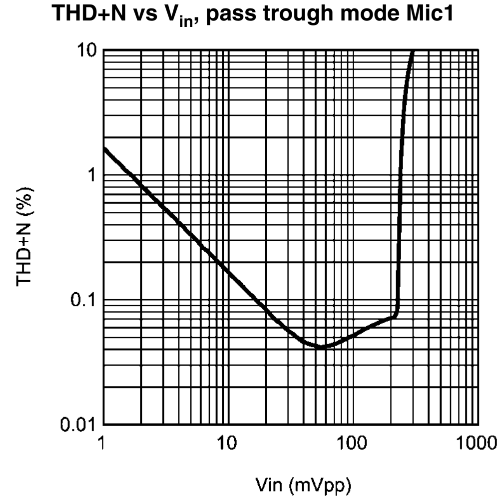
20213005

THD+N vs Frequency, Noise cancelling mode Mic1 AC shorted, signal at Mic2, V_{IN} = 36mV_{pp}



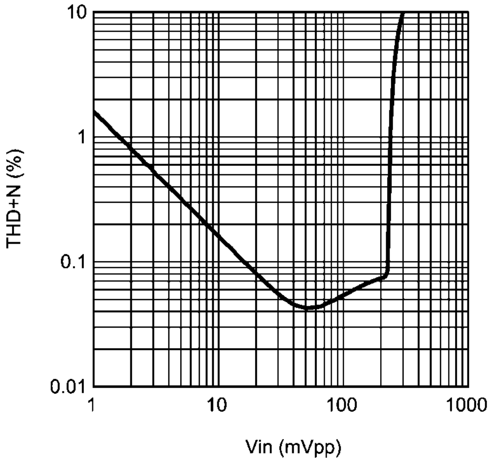
20213006

THD+N vs V_{in}, pass trough mode Mic1



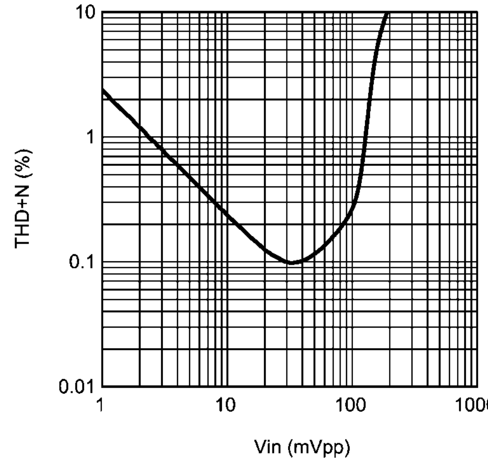
20213016

THD+N vs V_{in}, pass trough mode Mic2



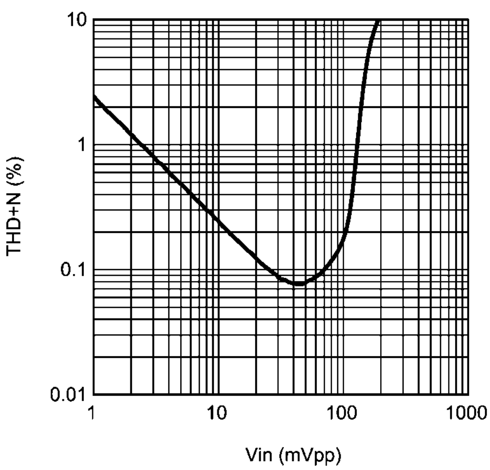
20213015

THD+N vs V_{in}, Noise canceling mode signal at Mic1, Mic2 AC shorted



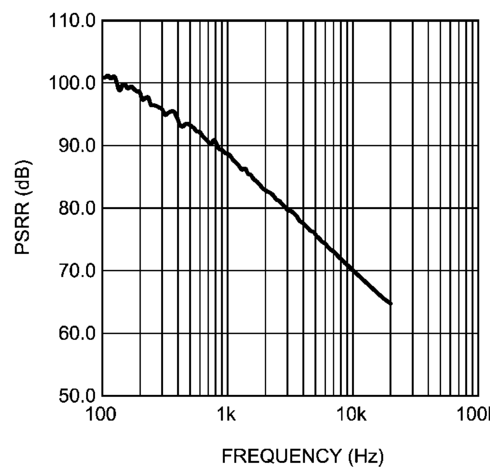
20213014

THD+N vs V_{in}, Noise canceling mode Mic1 AC shorted, signal at Mic2



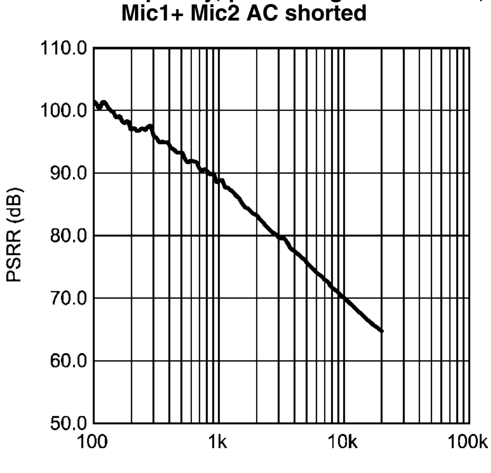
20213017

PSRR vs Frequency, pass trough mode Mic1, Mic1+ Mic2 AC shorted



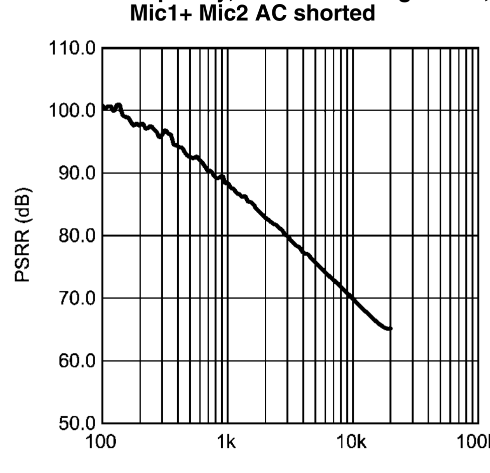
20213018

PSRR vs Frequency, pass trough mode Mic2, Mic1+ Mic2 AC shorted



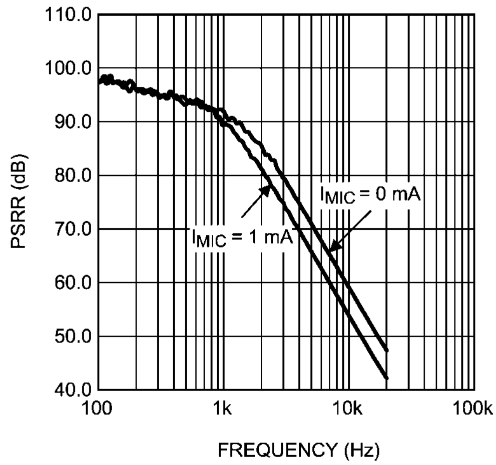
20213019

PSRR vs Frequency, Noise canceling mode , Mic1+ Mic2 AC shorted



20213020

PSRR vs Frequency, Microphone Bias , Mic1+ Mic2 AC shorted



20213022

Application Data

I²C Compatible Interface

I²C SIGNALS

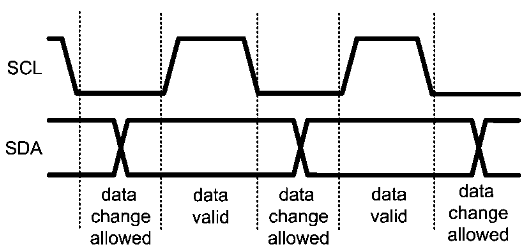
The LMV1088 pin SCL is used for the I²C clock SCL and the pin SDA is used for the I²C data signal SDA. Both these signals need a pull-up resistor according to I²C specification. The LMV1088 can be controlled on two slave addresses depending on the logical level at the I²C address pin. The two I²C slave address for LMV1088 are given in *Table 2* .

TABLE 2. Chip Address

	D7	D6	D5	D4	D3	D2	D1	D0
1st Chip Address I²C Adress='0'	1	1	0	0	1	1	0	$\overline{W}$/R
2nd Chip Address I²C Adress='1'	1	1	0	0	1	1	1	$\overline{W}$/R

I²C DATA VALIDITY

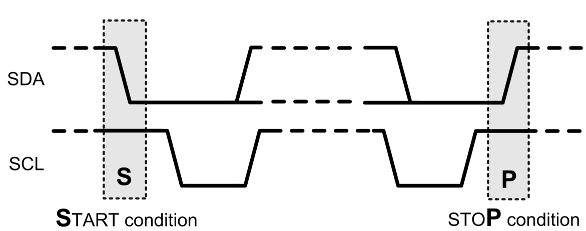
The data on SDA line must be stable during the HIGH period of the clock signal (SCL). In other words, state of the data line can only be changed when SCL is LOW.



I²C Signals: Data Validity

I²C START AND STOP CONDITIONS

START and STOP bits classify the beginning and the end of the I²C session. START condition is defined as SDA signal transitioning from HIGH to LOW while SCL line is HIGH. STOP condition is defined as the SDA transitioning from LOW to HIGH while SCL is HIGH. The I²C master always generates START and STOP bits. The I²C bus is considered to be busy after START condition and free after STOP condition. During data transmission, I²C master can generate repeated START conditions. First START and repeated START conditions are equivalent, function-wise.(Note 13)



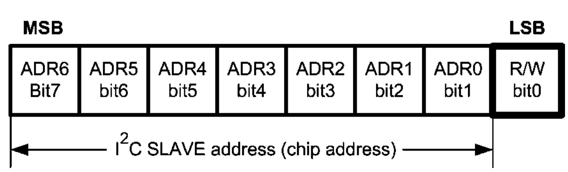
I²C Start Stop Conditions

Note 13: The master should issue STOP after no acknowledgement.

TRANSFERRING DATA

Every byte put on the SDA line must be eight bits long, with the most significant bit (MSB) being transferred first. Each byte of data has to be followed by an acknowledge bit. The acknowledge related clock pulse is generated by the master. The transmitter releases the SDA line (HIGH) during the acknowledge clock pulse. The receiver must pull down the SDA line during the 9th clock pulse, signifying an acknowledge. A receiver which has been addressed must generate an acknowledge after each byte has been received.

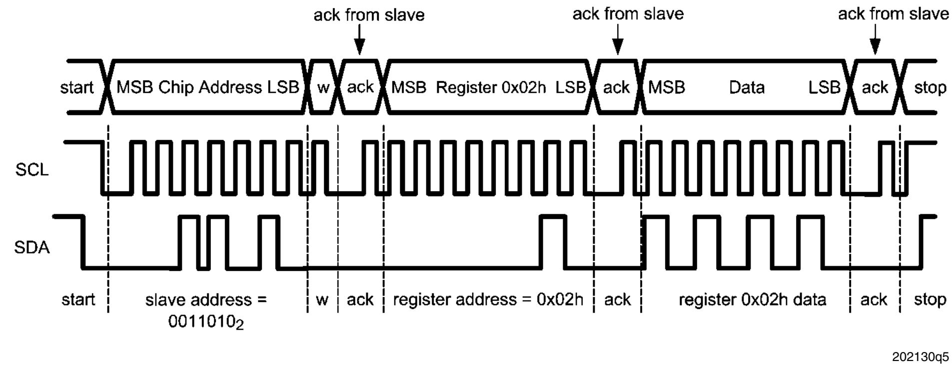
After the START condition, the I²C master sends a chip address. This address is seven bits long followed by an eighth bit which is a data direction bit (R/W). The LMV1088 address is **11001100_2**or **11001110_2**. For the eighth bit, a “0” indicates a WRITE and a “1” indicates a READ. The second byte selects the register to which the data will be written. The third byte contains data to write to the selected register.



I²C Chip Address

Register changes take effect at the SCL rising edge during the last ACK from slave.

In *Figure 2* there is a write example shown, for a device at a random chosen address'**00110100_2**'.



w = write (SDA = "0")
r = read (SDA = "1")
ack = acknowledge (SDA pulled down by slave)
rs = repeated start

FIGURE 2. Example I²C Write Cycle

When a READ function is to be accomplished, a WRITE function must precede the READ function, as shown in the Read Cycle waveform.

In Figure 3, there is a read example shown, for a device at a random chosen address'**00110101_2**'.

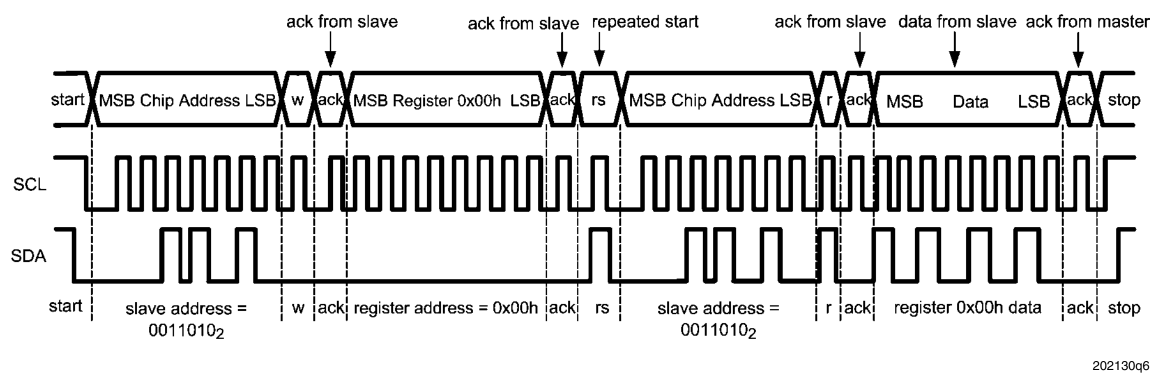


FIGURE 3. Example I²C Read Cycle

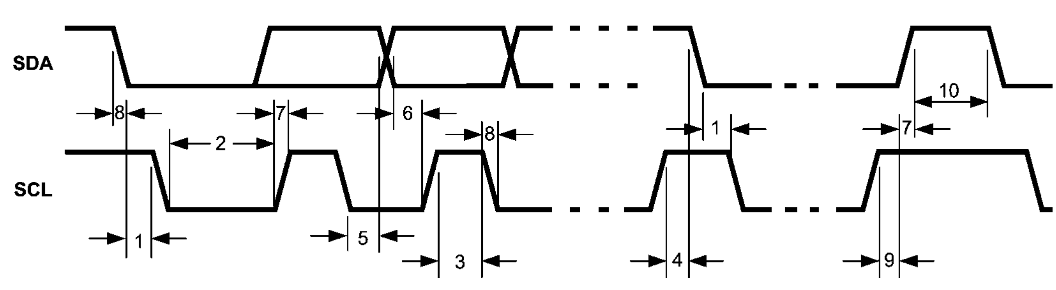


FIGURE 4. I²C Timing Diagram

TABLE 3. I²C Timing Paramters

Symbol	Parameter	Limit Min	Limit Max	Units
1	Hold Time (repeated) START Condition	0.6		μs
2	Clock Low Time	1.3		μs
3	Clock High Time	600		ns
4	Setup Time for a Repeated START Condition	600		ns
5	Data Hold Time (Output direction, delay generated by LMV1088)	300	900	ns
5	Data Hold Time (Input direction, delay generated by the Master)	0	900	ns
6	Data Setup Time	100		ns
7	Rise Time of SDA and SCL	20	300	ns
8	Fall Time of SDA and SCL	15	300	ns
9	Set-up Time for STOP condition	600		ns
10	Bus Free Time between a STOP and a START Condition	1.3		μs
C_b	Capacitive Load for Each Bus Line	10	200	pF

NOTE: Data guaranteed by design

TABLE 4. Register Map

Address	Reg.	7	6	5	4	3	2	1	0
0x01h	A	Men[2]	Men[1]	M[2]	M[1]	MPA[3]	MPA[2]	MPA[1]	MPA[0]
0x02h	B	0	0	0	MicSel[1]	MicSel[0]	Gpa[2]	Gpa[1]	Gpa[0]
0x12h	R	0	0	0	0	0	0	0	CAL

TABLE 5. I²C Register Description

Reg.	Bits	Description			Default
A	[3:0]	Microphone preamplifier gain from 6dB up to 36dB in 2dB steps.			0111
		0000	6dB		
		0001	8dB		
		0010	10dB		
		0011	12dB		
		0100	14dB		
		0101	16dB		
		0110	18dB		
		0111	20dB	default	
		1000	22dB		
		1001	24dB		
		1010	26dB		
		1011	28dB		
		1100	30dB		
		1101	32dB		
		1110	34dB		
		1111	36dB		
A	[5:4]	A4 = Mute mic1 and A 5 = mute mic2. (0 = microphone on)			00(on)
A	[7:6]	Mic enable bits, A6 = enable Mic1, A7 = enable Mic2 (1 = enable)			11(on)

<table>
<tr><th>Reg.</th><th>Bits</th><th colspan="4">Description</th><th>Default</th></tr>
<tr><td rowspan="10">B</td><td rowspan="10">[2:0]</td><td colspan="4">Gain setting for the post amplifier from (3dB steps) (Note 11).</td><td rowspan="10">000</td></tr>
<tr><td></td><td>Pass Through mode</td><td>Noise Canceling mode</td><td></td></tr>
<tr><td>000</td><td>-2.5dB</td><td>0db</td><td>default</td></tr>
<tr><td>001</td><td>0.5dB</td><td>3dB</td><td></td></tr>
<tr><td>010</td><td>3.5dB</td><td>6dB</td><td></td></tr>
<tr><td>011</td><td>6.5dB</td><td>9dB</td><td></td></tr>
<tr><td>100</td><td>9.5dB</td><td>12dB</td><td></td></tr>
<tr><td>101</td><td>9.5dB</td><td>12dB</td><td></td></tr>
<tr><td>110</td><td>9.5dB</td><td>12dB</td><td></td></tr>
<tr><td>111</td><td>9.5dB</td><td>12dB</td><td></td></tr>
<tr><td rowspan="5">B</td><td rowspan="5">[4:3]</td><td colspan="4">Mic select bits</td><td rowspan="5">00</td></tr>
<tr><td>00</td><td></td><td colspan="2">Noise canceling mode</td></tr>
<tr><td>01</td><td></td><td colspan="2">Only Mic1 on</td></tr>
<tr><td>10</td><td></td><td colspan="2">Only Mic2 on</td></tr>
<tr><td>11</td><td></td><td colspan="2">Mic1 + Mic2</td></tr>
<tr><td>B</td><td>[7:5]</td><td colspan="4">Not Used</td><td>000</td></tr>
<tr><td>R</td><td>[0]</td><td colspan="4">Start Calibration via I^2C
'0' to '1'= start calibration (keep '1' during calibration)</td><td>0</td></tr>
<tr><td>R</td><td>[7:1]</td><td colspan="4">internal test</td><td>0000000</td></tr>
</table>

Calibration

The full automatic calibration should only be required once, when the product containing the LMV1088 has completed manufacture, and prior to application packaging. The product containing the LMV1088 will be calibrated to the microphones, the microphone spacings, and the acoustical properties of the final manufactured product containing the LMV1088.

The compensation or calibration technology is achieved via memory stored coefficients when the FFNS circuitry activates the calibration sequence. The purpose of the calibration sequence is to choose the optimized coefficients for the FFNS circuitry for the given microphones, spacing, and acoustical environment of the product containing the LMV1088

A basic calibration can be performed with a single 1kHz tone, however to take full advantage of this calibration feature a three tone calibration (See the section *PERFORMING A THREE TONE CALIBRATION*) is preferred .

The automatic calibration process can be initiated from either a digital interface CALIBRATE pin (CAL) or via the I²C interface.

The logic level at the PROGRAM ENABLE (PE) pin determines if the result of the calibration is volatile or permanent. To make the result of the calibration permanent (stored in the EEPROM) the PROGRAM ENABLE (PE) pin must be high during the automatic calibration process.

AUTOMATIC CALIBRATION VIA I²C COMMAND

To initiate the automatic calibration via the CAL pin, the following procedure is required:

- From the initial condition where both PE and CAL are at 'low' level
- bring PE to a 'high' level (enable EEprom write)
- bring CAL to a 'high' level to start Calibration
- Apply Audio stimulus (single tone 1kHz or three tone sequence as described in *PERFORMING A THREE TONE CALIBRATION*)
- Hold CAL 'high' for at least 770ms
- Remove Audio stimulus
- bring CAL to a 'low' level to stop Calibration
- bring PE to a 'low' level (disable EEprom write)

A tone may be applied prior to the rising of CAL and PE. Signals applied to the microphone inputs before rising of CAL and PE are ignored by the calibration system.

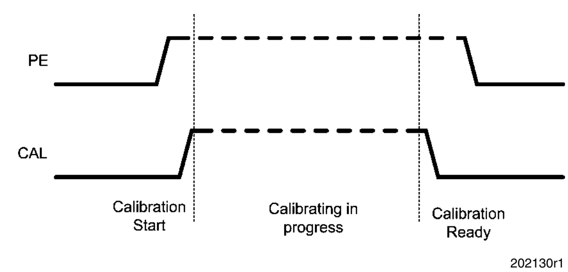


FIGURE 5. Automatic Calibration via CAL pin

Note: When the I²C is operated, make sure that register 'R' (address 0x12) bit 0 is '0' before operating the CAL pin (default value for this bit). When this bit is set '1' the calibration engine of the LMV1088 is started and will remain active with a higher supply current than normal operation. The state of the calibration remains active until this bit is reset, '0". With the bit set the 'low' to' high' transfer of the CAL pin will be ignored.

AUTOMATIC CALIBRATION VIA CAL PIN

To initiate the automatic calibration via the I² interface, the following procedure is required:

- From the initial condition where PE is 'low' level
- Bring PE to a 'high' level (enable EEprom write)
- Write '1' into I²C register 'R' (address 0x12) bit 0 to start calibration
- Apply Audio stimulus (single tone 1kHz or three tone sequence as described in *PERFORMING A THREE TONE CALIBRATION*)
- Wait at least 770ms
- Remove Audio stimulus
- Write '0' into I²C to finish calibration
- Bring PE to a 'low' level (disable EEprom write)

A tone may be applied prior to the rising of CAL and PE. Signals applied to the microphone inputs before rising of CAL and PE are ignored by the calibration system. .

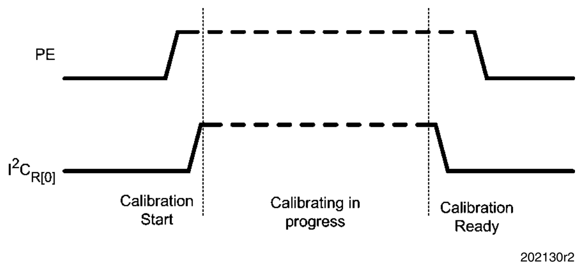


FIGURE 6.

PERFORMING A THREE TONE CALIBRATION

In a system with two microphones in an enclosure there will always be a difference in the transfer function in both gain and frequency response. The LMV1088 has the capability to perform an automatic calibration function to minimize these differences. To perform this calibration, a test sequence of three tones is required right after the PE and CAL inputs are brought to a logic high level. At the end of this sequence the calibration data is automatically stored in the internal EEPROM.

The three tones have to be applied as follows:

- A first tone with a frequency of 1kHz
- A second tone with a frequency of 300Hz
- A third tone with a frequency of 3kHz

A tone may be applied prior to the rising of CAL and PE. Signals applied to the microphone inputs before rising of CAL and PE are ignored by the calibration system. .

Between each tone pair there is a small time, indicated by a cross, to change the frequency. During that time the input tone is ignored by the calibration system.

The total calibration sequence requires less then 770ms.

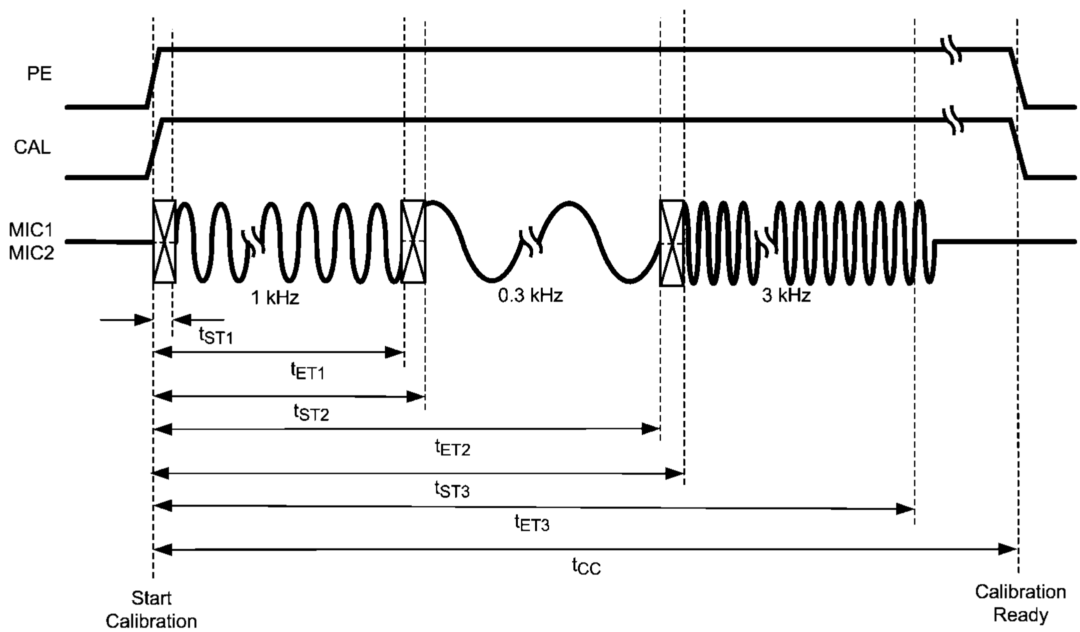


FIGURE 7. Three Tone Calibration Timing

TABLE 6. Automatic Calibration Timing Parameters

Symbol	Parameter	Limits		Unitis
		Min	Max	
t_{ST1}	Calibration Start Tone 1		10	ms
t_{ET1}	Calibration End Tone 1	200		ms
t_{ST2}	Calibration Start Tone 2		215	ms
t_{ET2}	Calibration End Tone 2	400		ms
t_{ST3}	Calibration Start Tone 3		415	ms
t_{ET3}	Calibration End Tone 3	600		ms
t_{CC}	Calibration Complete	770		ms

NOTE: Data guaranteed by design

THREE TONE CALIBRATION SETUP

A calibration test setup consist of a test room (acoustical box) with a loudspreaker (acoustical source) driven with the test tone sequence from *Figure 7*. The test setup is shown in *Figure 8*. The distance between the source and microphone 1 and microphone 2 must be equal and the sound must travel without any obstacle from source to both microphones.

The sound will travel with the limited speed of 300m/s from the loudspeaker source to the microphones. When creating the calibration signals this time should not be ignored, 30cm distance will cause 1ms delay.

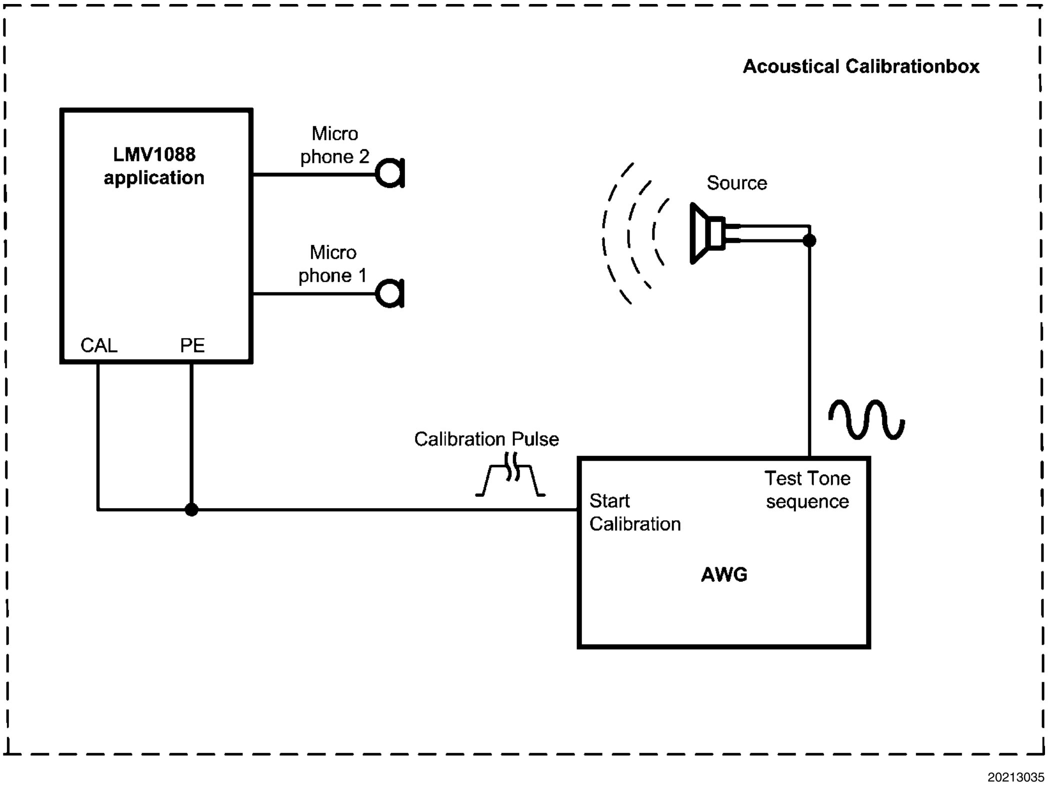


FIGURE 8. Three Tone Calibration Test setup

SUPPLY CURRENT DURING CALIBRATION

The Calibration function performs two main tasks in a sequence. First the AC characteristics of the microphones are matched. Then in the second stage, if the PE pin is high, the on-chip EEPROM is programmed.

During the first stage of this sequence the supply current on the LMV1088 will increase to about 2.5 mA. During the writing of the EEPROM the supply current will rise for about 215ms to about 30 mA. This increased current is used for the on chip charge pump which generates the high voltages that are required for programming the EEPROM.

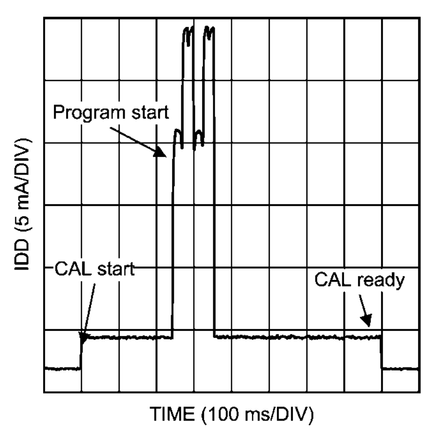


FIGURE 9. Supply current during calibration and programming

Low-Pass Filter At The Output

At the output of the LMV1088 there is a provision to create a 1st order low-pass filter (only enabled in 'Noise Cancelling' mode). This low-pass filter can be used to compensate for the change in frequency response that results from the noise cancellation process.. The change in frequency response resembles a first-order high-pass filter, and for many of the applications it can be approximately compensated by a first-order low-pass filter with cutoff frequency between 1.5kHz and 2.5kHz.

The transfer function of the low pass filter is derived as:

$$H(s) = \frac{R_{if}}{R_{if}C_f+1}$$

This low-pass filter is created by connecting a capacitor between the LPF pin and the OUT pin of the LMV1088. The value of this capacitor also depends on the selected output gain. For different gains the feedback resistance in the Low-pass Filter network changes as shown in *Table 7*.

TABLE 7. Low-pass Filter internal impedance

Post Amplifier Gain Setting (dB) (Note 14)	**Feedback resistance R_{if} (kΩ)**
0	20
3	29
6	40
9	57
12	80

This will result in the following values for a cutoff frequency of 2000 Hz:

TABLE 8. Low—pass Filter Capacitor for 2kHz

Post Amplifier Gain Setting (dB) (Note 14)	**R_{if} (kΩ)**	**C_f (nF)**
0	20	3.9
3	29	2.7
6	40	2.0
9	57	1.3
12	80	1.0

Note 14: Noise Cancelling Mode

Measurement Setup

Because of the nature of the calibration system it is not possible to predict the absolute gain in the two microphone channels of the Far Field Noise Cancelling System. This is because, after the calibration function has been operated, the noise cancelling circuit will compensate for the difference in gain between the microphones. In Noise Cancelling mode, this can result in a final gain offset of max 3dB between the gain set in the registers (RA[3:0] and RB[2:0]) and the actual measured gain between input and output of the LMV1088. After performing a calibration the frequency characteristic of the microphone channels will be matched for the two microphones. As a result of this matching there can be a slight slope in the frequency characteristic in one or both amplifiers.

A-WEIGHTED FILTER

The human ear is sensitive for acoustic signals within a frequency range from about 20Hz to 20kHz. Within this range the sensitivity of the human ear is not equal for each frequency. To approach the hearing response, weighting filters are introduced. One of those filters is the A-weighted filter.

The A-weighted filter is used in signal to noise measurements and THD+N measurements, where the wanted audio signal is compared to device noise and distortion.

The use of this filter improves the correlation of the measured values to the way these ratios are perceived by the human ear.

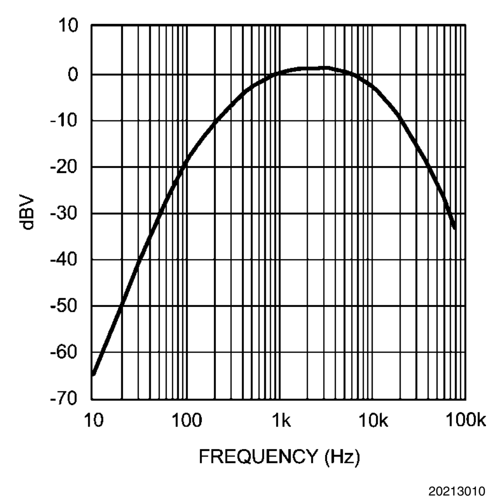


FIGURE 10. A-Weighted Filter

MEASURING NOISE AND SNR

The overall noise of the LMV1088 is measured within the frequency band from 10Hz to 22kHz using an A-weighted filter.

The Mic+ and Mic- inputs of the LMV1088 are shorted for AC signals via a short between the input capacitors , see *Figure 11*.

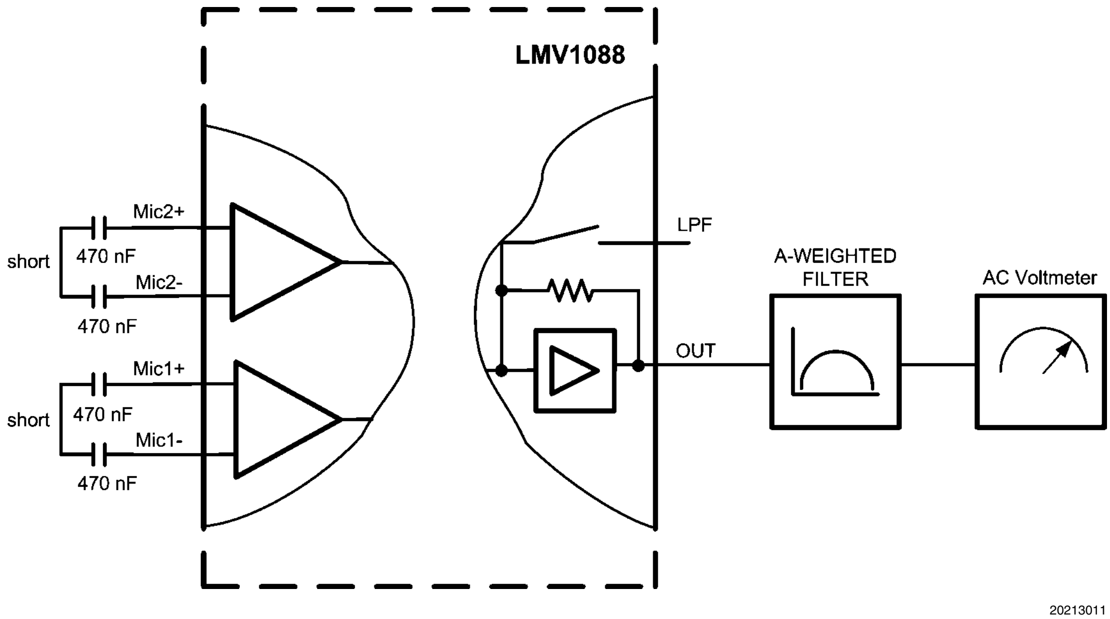


FIGURE 11. Noise Measurement Setup

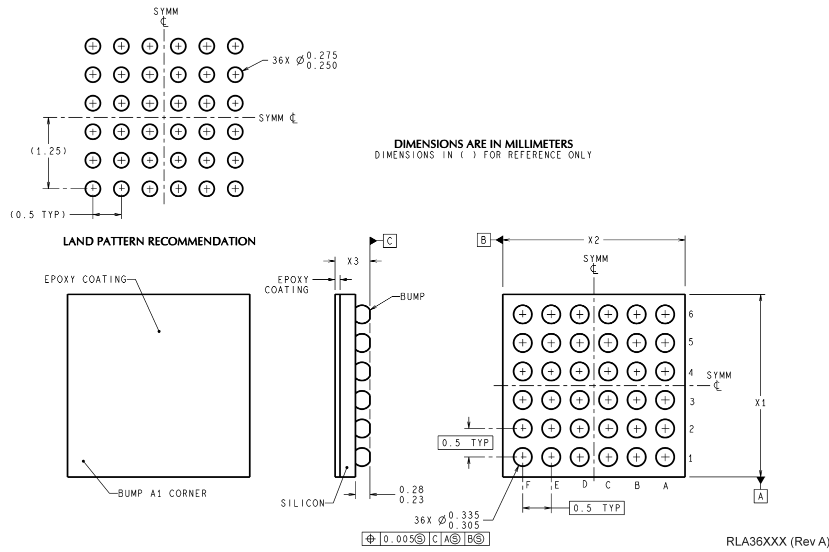
For the signal to noise ratio (SNR) the signal level at the output is measured with a 1kHz input signal of 18mV$_{PP}$ using an A-weighted filter. This voltage represents the output voltage of a typical electret condenser microphone at sound pressure level of 94dB SPL, which is the standard level for these measurements. The LMV1088 is programmed for 17.5dB of total gain (20dB pre-amplifier and -2.5dB post-amplifier) with only Mic1 or Mic2 on. (See also *I²C Compatible Interface*)

The input signal is applied differential between the corresponding Mic+ and Mic- . Because the part is in Pass Through mode the Low-pass Filter at the output of the LMV1088 is disabled.

Revision History

Rev	Date	Description
1.0	09/26/07	Initial release.
1.01	12/10/07	Few text edits (changed TL to RL).

Physical Dimensions inches (millimeters) unless otherwise noted



36 Bump micro SMD Technology
NS Package Number RLA36VVA
X_1 = 3.51±0.03, X_2 = 3.51±0.03, X_3 = 0.6±0.075,

Notes

For more National Semiconductor product information and proven design tools, visit the following Web sites at:

Products		Design Support	
Amplifiers	www.national.com/amplifiers	WEBENCH	www.national.com/webench
Audio	www.national.com/audio	Analog University	www.national.com/AU
Clock Conditioners	www.national.com/timing	App Notes	www.national.com/appnotes
Data Converters	www.national.com/adc	Distributors	www.national.com/contacts
Displays	www.national.com/displays	Green Compliance	www.national.com/quality/green
Ethernet	www.national.com/ethernet	Packaging	www.national.com/packaging
Interface	www.national.com/interface	Quality and Reliability	www.national.com/quality
LVDS	www.national.com/lvds	Reference Designs	www.national.com/refdesigns
Power Management	www.national.com/power	Feedback	www.national.com/feedback
Switching Regulators	www.national.com/switchers		
LDOs	www.national.com/ldo		
LED Lighting	www.national.com/led		
PowerWise	www.national.com/powerwise		
Serial Digital Interface (SDI)	www.national.com/sdi		
Temperature Sensors	www.national.com/tempsensors		
Wireless (PLL/VCO)	www.national.com/wireless		

For the most current product information visit us at www.national.com

National Semiconductor Americas Technical Support Center
Email: new.feedback@nsc.com
Tel: 1-800-272-9959

National Semiconductor Europe Technical Support Center
Email: europe.support@nsc.com
German Tel: +49 (0) 180 5010 771
English Tel: +44 (0) 870 850 4288

National Semiconductor Asia Pacific Technical Support Center
Email: ap.support@nsc.com

National Semiconductor Japan Technical Support Center
Email: jpn.feedback@nsc.com