APPLICATION NOTE

Application of the TEA1099

Speech and Handsfree IC
with auxiliary inputs/output
and analog multiplexer

AN98061

Philips

Semiconductors

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Abstract

The TEA1099 is a bipolar circuit which includes line interface, speakerphone function and switches for connection of auxiliary interfaces.It is intended to be used in line or mains powered telephone terminals.

A detailed description of the circuit blocks of the TEA1099 and advices on adjustments are contained in this report.

APPLICATION NOTE

Application of the TEA1099 Speech and Handsfree IC with auxiliary inputs/output and analog multiplexer

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Keywords

Telecom
Demonstration Board
TEA1099
transmit
receive
auxiliary
line

Date: June 11th, 1998

Summary

A detailed description of the TEA1099 is given.

The TEA1099 incorporates a line interface block with microphone, earphone and DTMF amplifiers.

It incorporates also a duplex controller with signal and noise monitors on the transmit and receive channels, a base microphone amplifier as well as a loudspeaker amplifier.

In addition, two auxiliary inputs and one auxiliary output combined with integrated switches allow the use of the TEA1099 in a lot of applications which can be either line-powered or powered from the mains.

A cookbook gives the general application steps.

A demonstration board, OM5846, is available.

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1. INTRODUCTION

The TEA1099 is a circuit which offers a handsfree function with the line interface and the normal handset interface, it also incorporates auxiliary amplifiers combined with switches and a logic control block.

It incorporates a base microphone amplifier, a volume control of the loudspeaker amplifier and a duplex controller with signal and noise monitoring on the transmit and receive channels.

A power supply block extracts power from the line in an optimized way for the loudspeaker amplifier; furthermore, this supply block can be powered from any external supply. A stabilized 3.35 V supply is available for peripherals.

This makes the TEA1099 suitable as the core of a multifunction telecom terminal, such as cordless telephones, answering machines or fax machines.

The function of the handsfree application is illustrated with the help of fig.1.

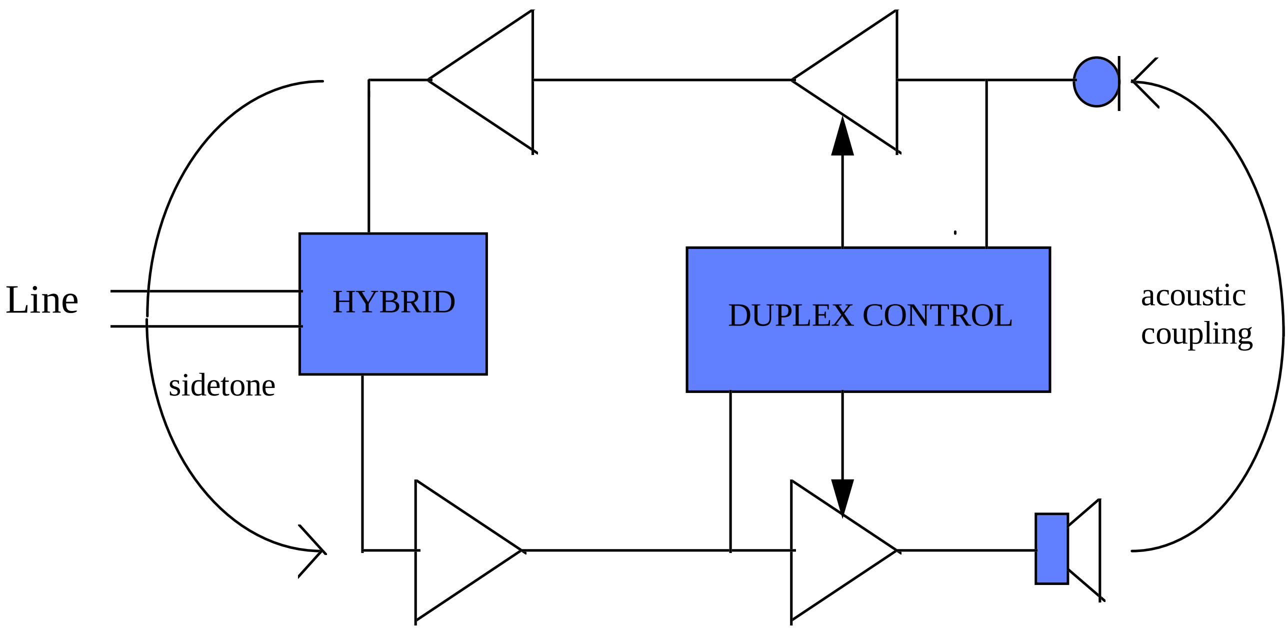

Fig. 1 Handsfree telephone set principle

The left side of fig. 1 shows a principle diagram of the line interface part of the TEA1099 circuit by means of a receiving preamplifier, a transmit amplifier and the hybrid. The right side of fig.1 shows a principle diagram of the handsfree part of the TEA1099 circuit by means of the microphone preamplifier, the loudspeaker amplifier and the duplex controller.

As can be seen from fig. 1, a closed loop is formed via the amplifiers, the antisidetone network and the acoustic coupling between loudspeaker and microphone. When the loop-gain is higher than one, the set starts howling. In a full-duplex application, this would be the case. To avoid howling, the duplex controller reduces the loop-gain to a value much lower than one.

The duplex controller of the TEA1099 monitors the signal and noise on both the transmit and the receive channel in order to detect which channel contains the 'largest' signal. As a result, the duplex controller reduces the gain of the channel which contains the smallest signal. The sum of the transmit and the receive gains remains constant.

As a result, the circuit can be in three stable modes to be referred to throughout this report:

1. Transmit mode (Tx-mode): the gain of the handsfree microphone path is at its maximum and the gain of the receive path (to loudspeaker amplifier) is reduced.
2. Receive mode (Rx-mode): the gain of the receive path is at its maximum and the gain of the microphone amplifier is reduced.
3. Idle mode (Ix-mode): the gains of the handsfree microphone path and of the receive path are halfway their maximum and reduced values.

The difference between the maximum gain and the reduced gain is called the switching range.

This report gives a detailed description of the TEA1099 and its basic application. The description is given by means of the block diagram of the TEA1099 (§2) and by discussing every detail of the sub-blocks (§3). The application is discussed by giving a guideline for application (the application cookbook §4) and by giving an application example (§5). EMC aspects are also discussed (§6). The appendix contains a list of abbreviations and the demoboard application diagram of the TEA1099.

Note: the values of parameters given in this application note are as accurate as possible, but please, refer to the last product specification for final ones.

2. BLOCK DIAGRAM

In this chapter, the block diagram of the TEA1099 is shown by means of fig. 2. The pinning of the TEA1099 is given by means of fig. 3. A short description of the block diagram is given including the function of the external components.

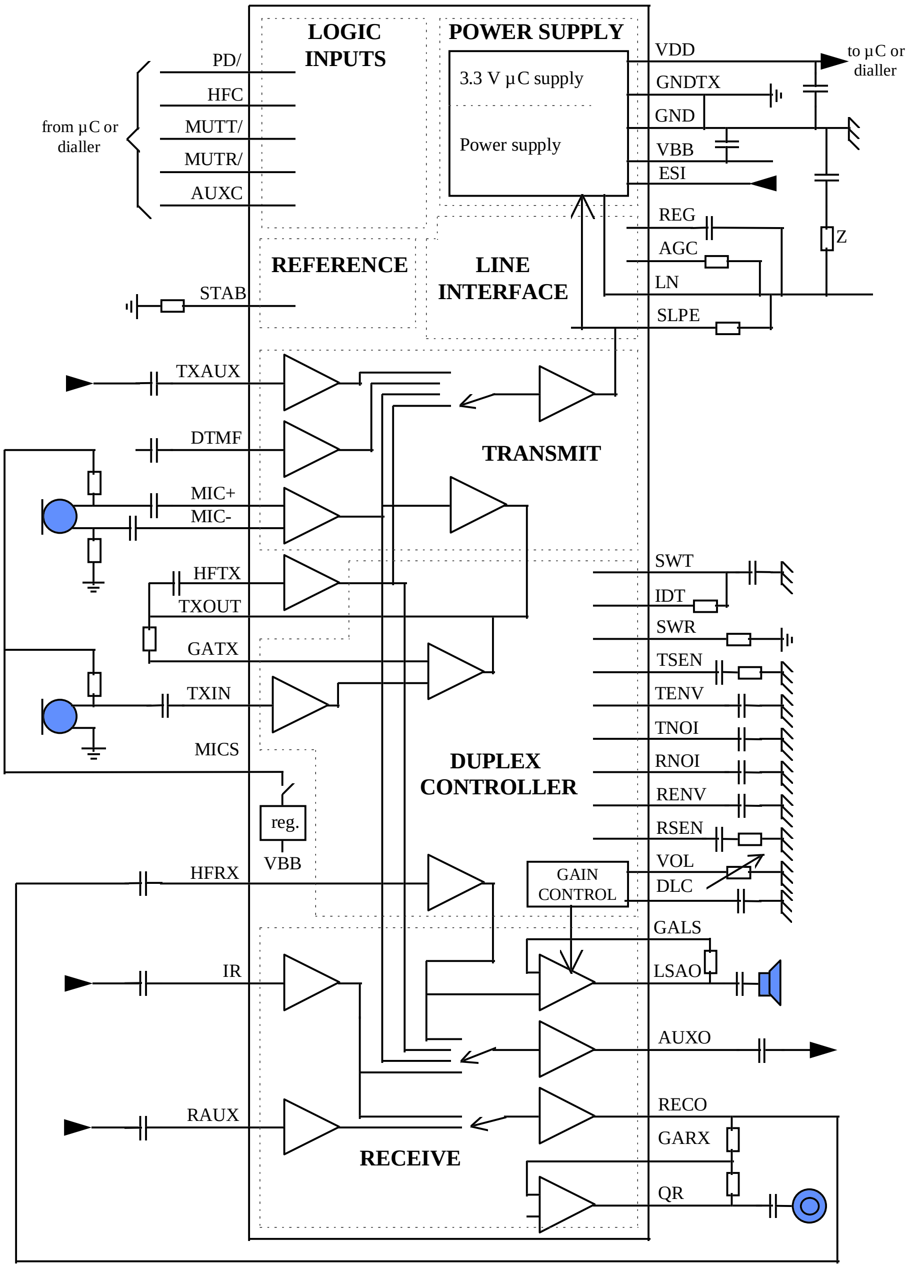

Fig. 2 Block diagram of TEA1099

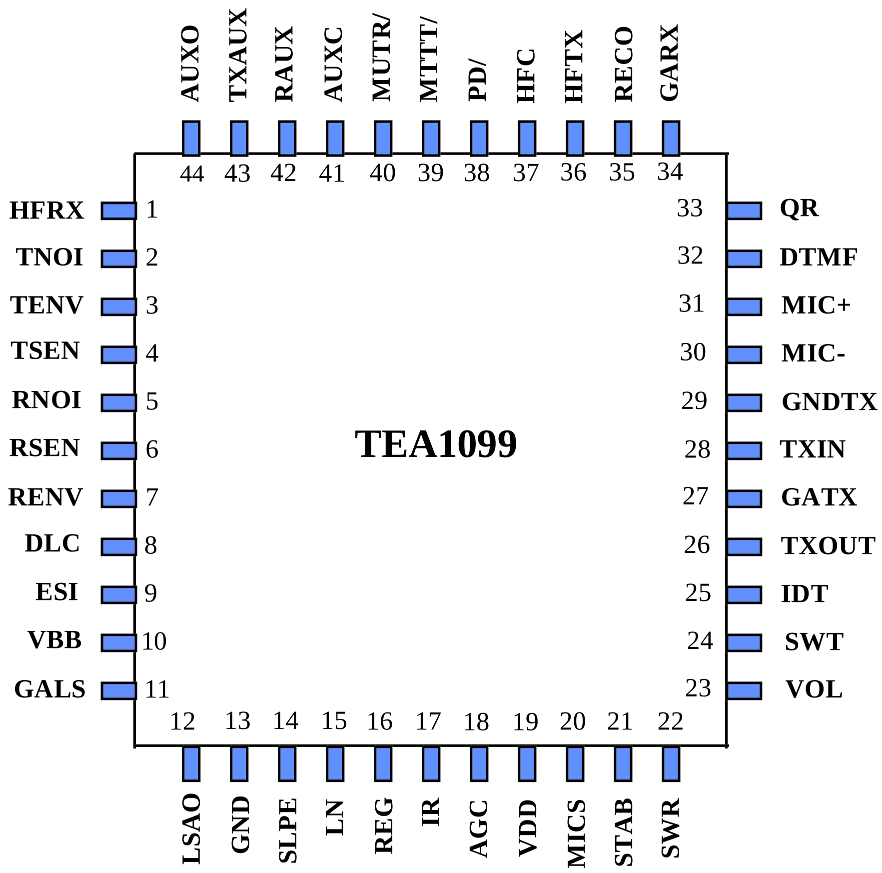

Fig. 3 Pinning of TEA1099

PIN	NAME	DESCRIPTION
1	HFRX	Receive input for loudspeaker amplifier or auxiliary receive amplifier
2	TNOI	Transmit noise envelope timing adjustment
3	TENV	Transmit signal envelope timing adjustment
4	TSEN	Transmit signal envelope sensitivity adjustment
5	RNOI	Receive noise envelope timing adjustment
6	RSEN	Receive signal envelope sensitivity adjustment
7	RENV	Receive signal envelope timing adjustment
8	DLC	Dynamic limiter
9	ESI	External supply input
10	VBB	Stabilized supply for internal circuitry
11	GALS	Loudspeaker amplifier gain adjustment
12	LSAO	Loudspeaker amplifier output
13	GND	Ground reference
14	SLPE	Line current sense

15	LN	Positive line terminal
16	REG	Line voltage regulator decoupling
17	IR	Receive channel input
18	AGC	Automatic gain control
19	VDD	3.35 V supply voltage
20	MICS	Supply for electret microphones
21	STAB	Reference current adjustment
22	SWR	Switching range adjustment
23	VOL	Loudspeaker amplifier volume adjustment
24	SWT	Switching timing adjustment
25	IDT	Idle-mode timing adjustment
26	TXOUT	Handsfree microphone amplifier output
27	GATX	Handsfree microphone gain adjustment
28	TXIN	Handsfree microphone amplifier input
29	GNDTX	Ground reference for microphone amplifiers
30	MIC-	Inverting HS microphone input
31	MIC+	Non-inverting HS microphone input
32	DTMF	Dual Tone Multifrequency input
33	QR	Earpiece amplifier output
34	GARX	Earpiece amplifier gain adjustment
35	RECO	Receive amplifier output
36	HFTX	Transmit input for line amplifier or auxiliary receive amplifier
37	HFC	Logic input
38	PD/	Power-down input
39	MUTT/	Logic input
40	MUTR/	Logic input
41	AUXC	Logic input
42	RAUX	Auxiliary receive input
43	TXAUX	Auxiliary transmit input
44	AUXO	Auxiliary output

In fig. 2 it can be seen that the IC consists out of six main parts: the line interface, the supply block, the transmit block, the receive block with the loudspeaker amplifier, the duplex controller and the logic block which controls the IC. These blocks are shortly described below including the function of the external components. The detailed description will follow in chapter 3.

Line interface :

The TEA1099 generates a stabilized voltage (called Vref) between pins GND and SLPE. This reference voltage is line current dependant in order to get optimum supply for the loudspeaker amplifier and is stabilized by the capacitor Creg connected at pin REG. The line current is sensed across the resistor connected between pins LN and SLPE.

An AGC function is provided when pin AGC is connected directly or through a resistor to LN.

The impedance of the apparatus is set by a network connected between LN and GND through a decoupling capacitor.

Supply:

The circuit can be supplied from the line and/or by an external supply. It provides a stabilized 3.35 V supply point for peripherals which can also be externally supplied in trickle mode. The TEA1099 can be switched into a low power consumption mode with the pin PD/.

Transmit:

The transmit signal can come from four preamplifiers: handset microphone (MIC+/MIC-), handsfree microphone (TXIN to TXOUT and HFTX), the auxiliary transmit (TXAUX) and the DTMF. The selection is made by the logic block.The signal reference is GNDTX, a “clean ground” which has to be connected to GND, for the handsfree microphone. The inputs have to be coupled by means of capacitors. All the gains have a fixed value except the gain of the handsfree microphone amplifier which is set with Rgatx.

Receive:

The signal received from the line is amplified from pin IR to pin RECO and/or to the auxiliary output AUXO. The input IR has to be coupled by means of a capacitor. From pin RECO, the signal is sent to the earphone amplifier at pins GARX and QR and to the input of the duplex controller HFRX. The gain of the earphone amplifier is set with 2 resistors. From HFRX, the signal can be sent to the loudspeaker amplifier (pins GALS and LSAO) and the volume can be adjusted by means of the potentiometer connected between input VOL and GNDTX, the gain of the loudspeaker amplifier is set by the resistor Rgals. The signal from the auxiliary input RAUX can also be sent to RECO.

Moreover the auxiliary output AUXO can get its signal from IR, HFRX through the duplex controller, MIC+/MIC- or HFTX.

Duplex controller:

From both the transmit and receive signals, signal and noise envelopes are made. The transmit signal envelope is on pin TENV and the receive one on pin RENV. The transmit noise envelope is on pin TNOI and the receive one on pin RNOI. The timing of the envelopes can be set by the capacitors Ctenv, Ctnoi, Crenv and Crnoi. The sensitivity of the envelope detectors can be set by means of the RC combinations Rtsen with Ctsen for the transmit envelope and Rrsen with Crsen for the receive one. The resistors set the sensitivity and the capacitors block the DC-component, creating also high-pass filters.

The decision logic of the duplex controller determines into which mode (Tx, Rx or Ix-mode) the set has to switch over. The timing for switching to the Tx or the Rx -mode is determined with the capacitor Cswt. The timing for switching to the Ix-mode is set by the combination Cswt and Ridt. The switching range is determined by the resistor Rswr. Resistor Rstab has a fixed value.

Logic block :

The logic block manages the internal switches according to the following table.

LOGIC INPUTS					CONNECTIONS	MODES
(PD/)	**HFC**	**MUTT/**	**MUTR/**	**AUXC**		
0	X	X	X	1	HFRX -> LSAO	Ring mode
0	X	X	X	0		Flash, DC dialling
1	0	0	0	0	DTMF->LN ; CT->RECO (MICS, QR)	Tel. Set: DTMF dialling
1	0	0	1	0	MIC->AUXO; RAUX->RECO (MICS, QR)	
1	0	0	1	1	MIC->AUXO; RAUX->RECO (QR)	
1	0	1	X	0	MIC->LN; IR->RECO; IR->AUXO MIC->TXOUT (MICS, QR)	Handset conversation
1	0	1	1	1	MIC -> LN; MIC -> TXOUT (QR)	
1	0	1	0	1	TXAUX->LN ; IR->AUXO	Conversation with auxiliary
1	0	0	0	1	TXAUX->LN ; IR->AUXO RAUX -> RECO	
1	1	1	1	1	TXIN->TXOUT; HFTX->LN; IR->RECO; HFRX->AUXO	
1	1	0	1	1	RAUX->RECO; HFRX->LSAO	
1	1	0	0	1	TXAUX->LN ; IR->AUXO ; RAUX->RECO; HFRX->LSAO	
1	1	0	0	0	DTMF->LN ; CT->RECO; HFRX->LSAO (MICS, QR)	HF/GL Tel Set DTMF dialling
1	1	1	0	1	TXAUX->LN ; IR->AUXO ; IR->RECO; HFRX->LSAO	
1	1	0	1	0	TXIN->TXOUT; HFTX->AUXO; RAUX->RECO; HFRX->LSAO (MICS)	
1	1	1	1	0	TXIN->TXOUT; HFTX->LN; IR->RECO; IR->AUXO; HFRX->LSAO (MICS)	Handsfree conversation
1	1	1	0	0	MIC->LN; IR->RECO; IR->AUXO; HFRX->LSAO MIC->TXOUT (MICS,QR)	Group-listenning conversation

Fig. 4 Table of switch management

3. DESCRIPTION OF THE TEA1099

This chapter describes in detail the six blocks of the speech/handsfree circuit TEA1099: the line interface (3.1), the supply (3.2), the transmit block (3.3), the receive block (3.4), the duplex controller (3.5) and the logic block (3.6). For each block the principle of operation is described and its adjustments and performances are discussed.

All values given in this chapter are typical and at room temperature unless otherwise stated. For more details, see TEA1099 device specification.

All the curves shown in this section result from measurement of typical samples using the schematic of fig. 66. All the component names refer to the basic application of the IC shown in appendix fig. 64.

3.1 Line interface

3.1.1 DC characteristics

Principle of operation

The TEA1099 generates a stabilized voltage (called Vref) between pins GND and SLPE. This reference voltage, temperature compensated, is typically 3.7 V for line currents between 10 and 18 mA and 6.15 V for line currents between 46 and 140 mA. For line currents between 18 and 46 mA, Vref increases proportionally to this line current with a slope of typically 87.5 Ω so, from typically 3.7 to 6.15 V. The voltage at pin REG is used by the internal regulator to generate the stabilized Vref voltage and is decoupled by a capacitor Creg connected to LN.

For effective operation of the apparatus, the TEA1099 must have a low resistance to the DC current and a high impedance to speech signals. The Creg capacitor, converted into an equivalent inductance (see “set impedance” section), realizes this impedance conversion from its DC value (Rslpe) to its AC value (Zimp in the audio frequency range). The DC voltage between pins LN and SLPE is proportional to the line current.

This general configuration is shown in fig. 5.

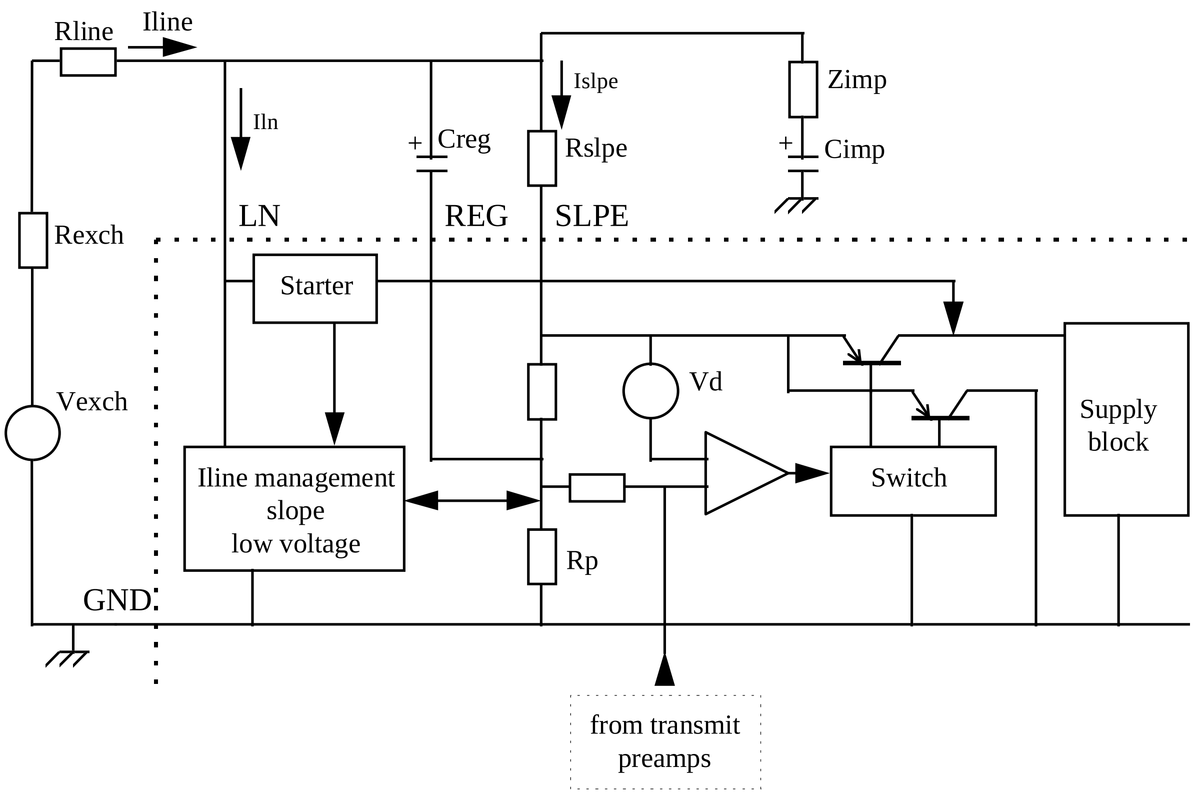

Fig. 5 DC characteristics configuration

The IC regulates the line voltage between pins GND and SLPE. The voltage on pin LN can be calculated as:

Vln = Vref + Rslpe × Islpe

Islpe = Iline - Iln

Iline = line current

Iln = current consumption between LN and GND

Between 18 and 46 mA:

Vref ≅ 3.7 + (Islpe - 18 mA) × 87.5

The DC line current Iline flowing into the apparatus is determined by the exchange supply voltage Vexch, the feeding bridge resistance Rexch, the DC resistance of the telephone line Rline and the voltage across the telephone set including diode bridge.

Below a threshold line current Ith (typically equal to 9 mA) the internal reference voltage (generating Vref) is automatically adjusted to a lower value (down to an absolute minimum voltage of 1.6 V). This means that more sets can operate in parallel or that for very low voltage feeding bridge the line current has a higher value. For line currents below this threshold current, the TEA1099 has reduced sending and receiving performances. This is called the low voltage area.

The internal circuitry of the TEA1099 is supplied from pin VBB. In line powered application, this voltage is derived from the line voltage by the supply block and must be decoupled by a capacitor (Cvbb). Fig. 6 shows the IC current consumption (Ibb) as a function of the VBB supply voltage in handset mode and in handsfree mode.

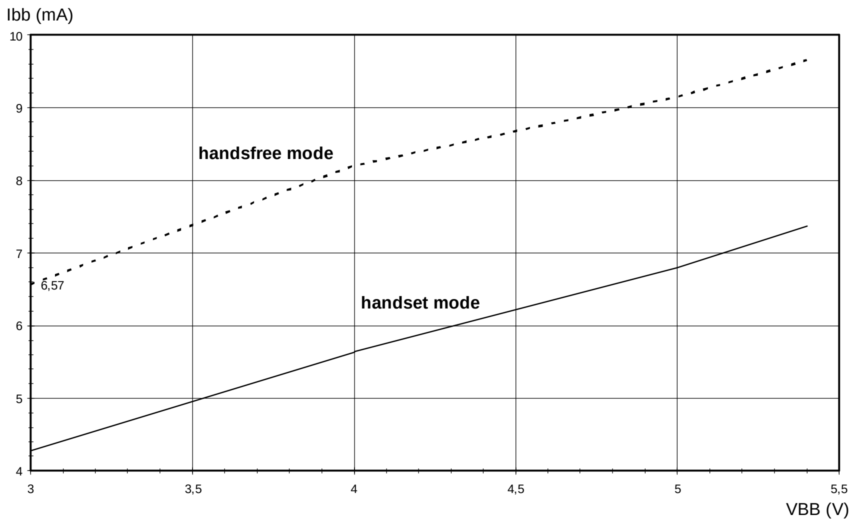

Fig. 6 Ibb versus Vbb

Fig. 7 shows the main voltages as a function of the line current in line powered conditions.

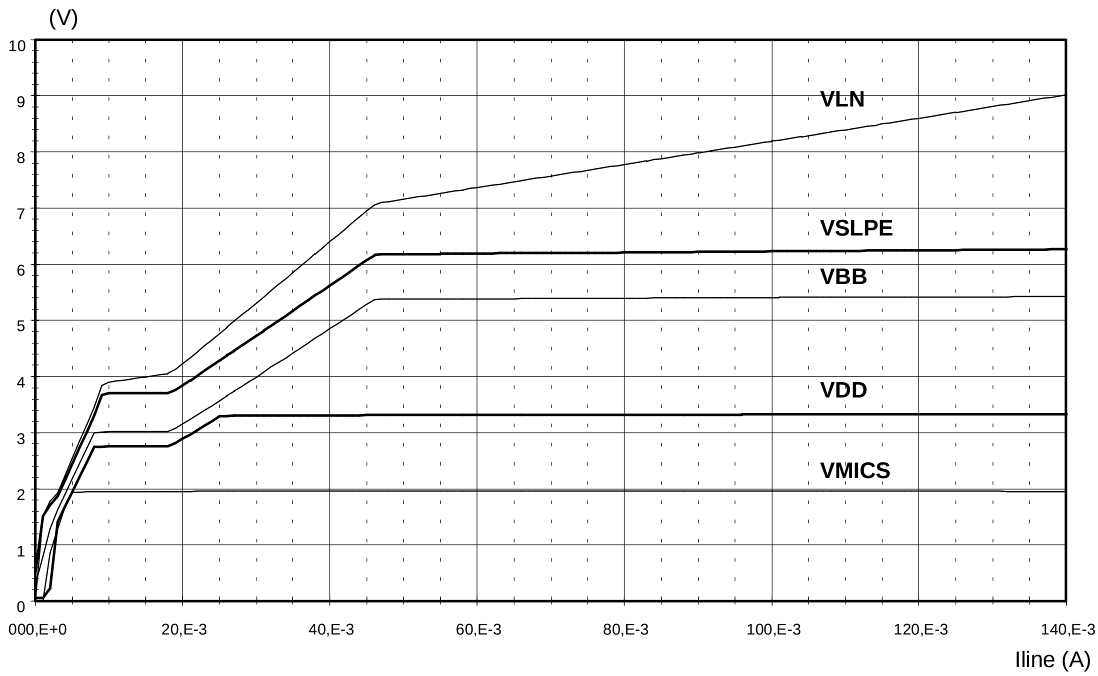

Fig. 7 Main voltages versus line current

Fig. 8 shows the behavior in the low voltage area in line powered condition.

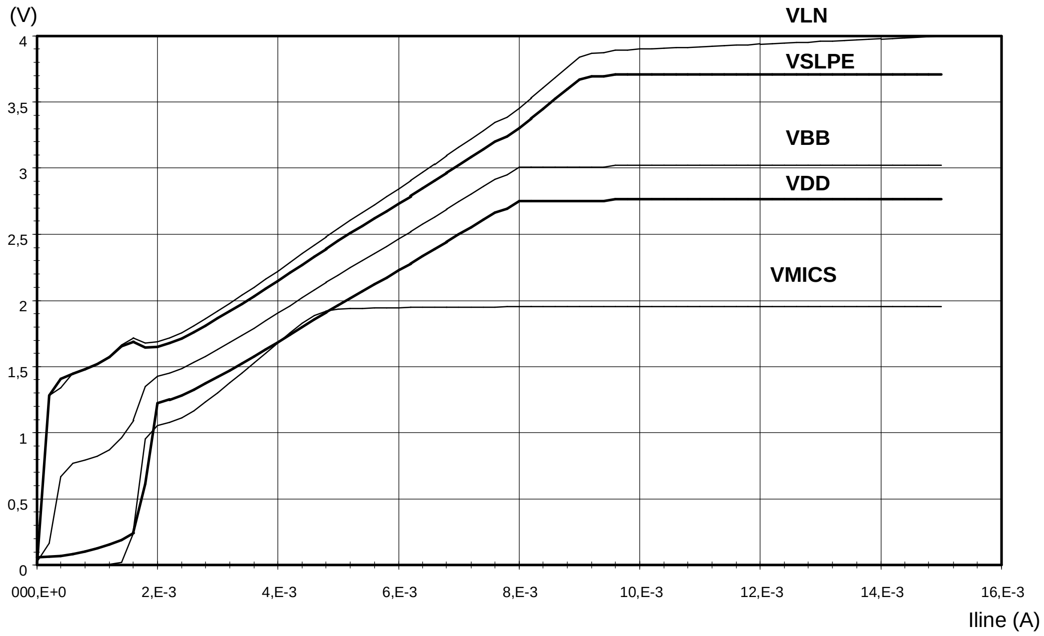

Fig. 8 Low voltage behavior in line powered condition

Fig. 9 shows the behavior in the low voltage area when VBB is externally supplied at 5 V.

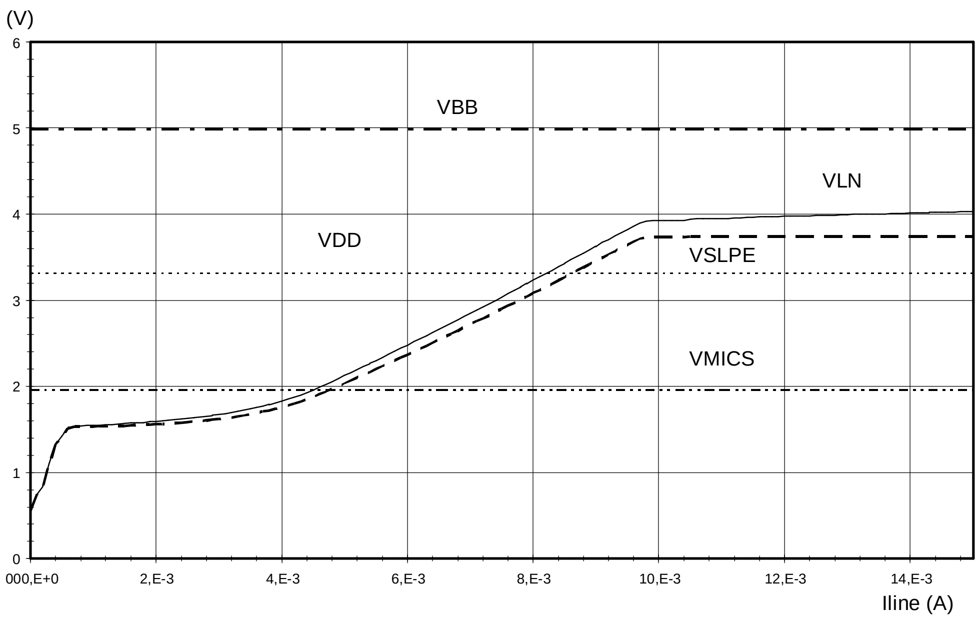

Fig. 9 Low voltage behavior when VBB = 5 V

Adjustments and performances

The reference voltage, Vref, can be adjusted by means of an external resistor Rva. It can be increased by connecting the Rva resistor between pins REG and SLPE (Fig. 10), or decreased by connecting the Rva resistor between pins REG and GND. In line powered application, it is possible to use the voltage reduction only for less than 300 mV because it reduces the VBB supply capability, this reduction is easier when VBB is provided by an external 5 V power supply. To ensure correct operation, it is not advised to adjust Vref at a value lower than 3.3 V at 18 mA or higher than 7 V at a maximum line current of 90 mA (the maximum operating voltage of 12 V must be guaranteed by the application as well as the safe crystal operating temperature). These adjustments will slightly affect a few parameters: there will be a small change in the temperature coefficient of Vref and a slight increase in the spread of this voltage reference due to matching between internal and external resistors. Furthermore, the Rva resistor connected between REG and GND will slightly affect the apparatus impedance (see section “set impedance”).

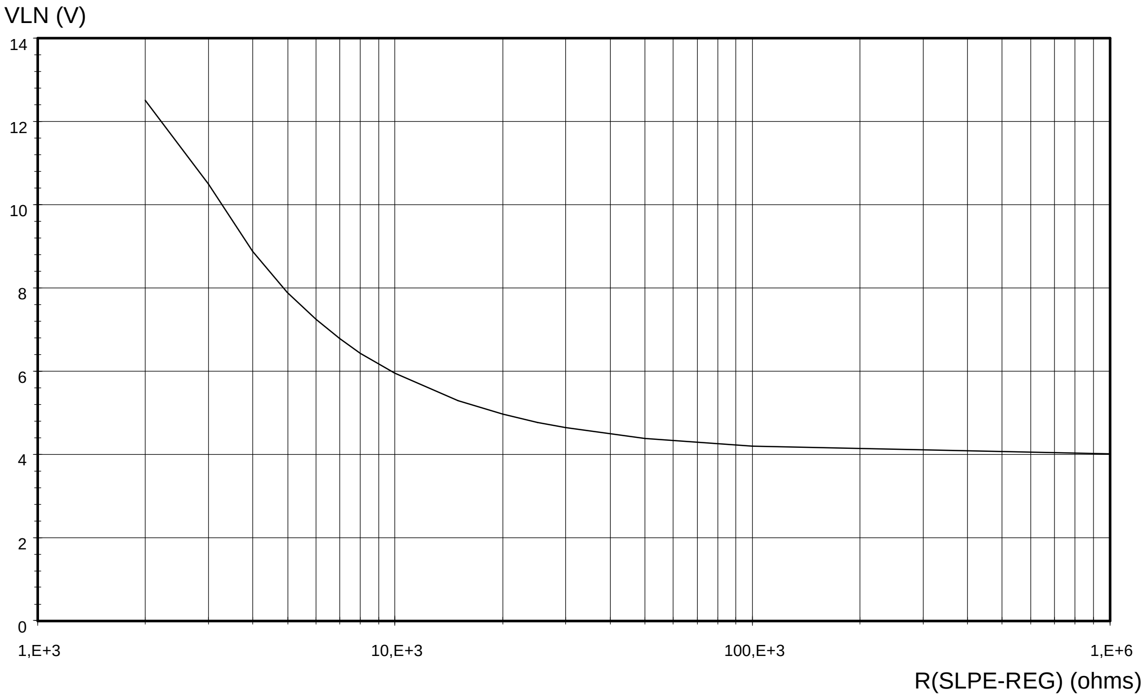

Fig. 10 Influence of the Rva resistor between REG and SLPE on Vln at 15 mA

The DC slope of the voltage on pin LN is influenced by the Rslpe resistor as shown in fig. 11. This value of Rslpe may be slightly modified even if the preferred one is 20 Ω; changing this value will affect more than the DC characteristics, it also influences the gains, the AGC characteristics, the maximum output swing on the line, the VBB slope start and stop currents and the low voltage threshold Ith.

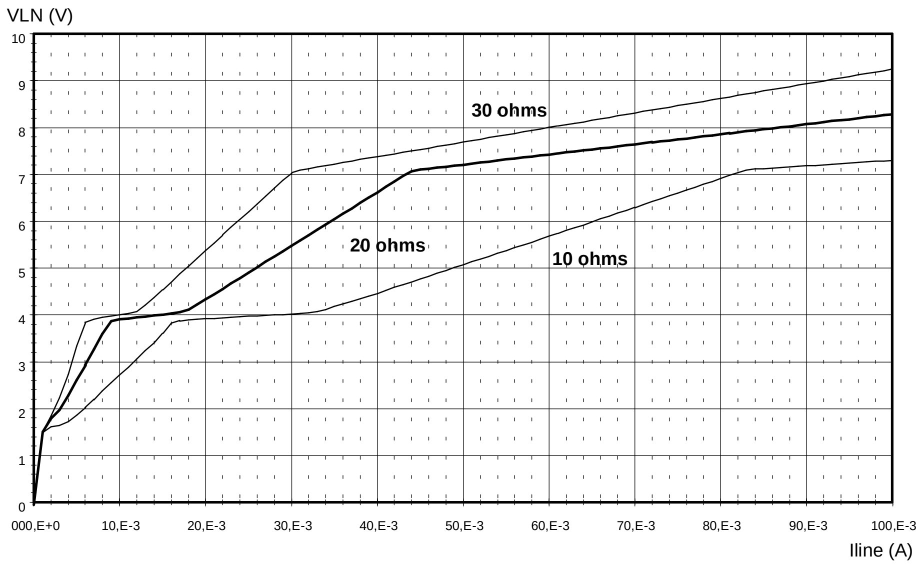

Fig. 11 Influence of Rslpe on the DC characteristics

3.1.2 Line impedance

Principle of operation

The TEA1099 behaves like an equivalent inductance that presents a low impedance to DC and a high impedance (Rp) to speech signals. Rp is an integrated resistance in the order of 25 kΩ +/-15%. It is in parallel with the external network realized by Zimp and Cimp. Thus, in the audio frequency range, the apparatus impedance (called set impedance) is mainly determined by the Zimp resistor. Fig. 12 shows an equivalent schematic for the set impedance.

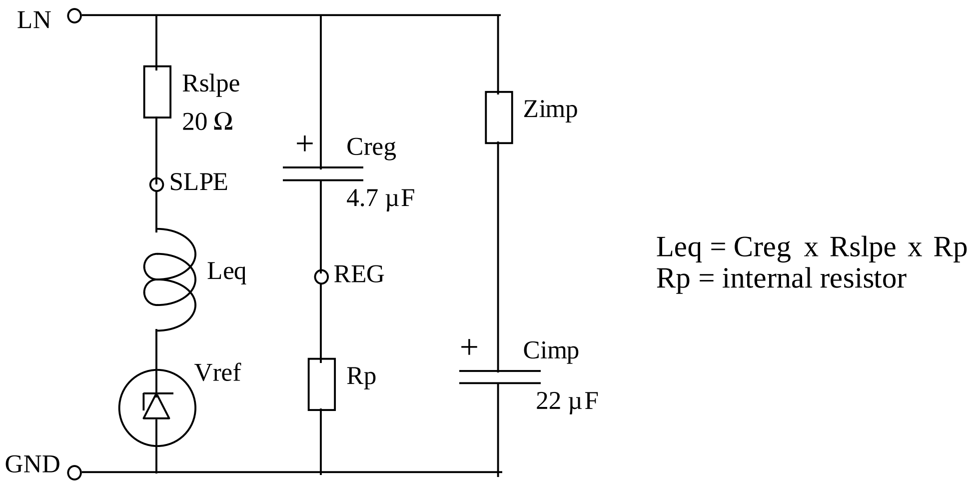

Fig. 12 Equivalent set impedance

Adjustments and performances

When decreasing the reference voltage Vref, a resistor is connected between GND and REG in parallel of Rp (see fig. 12) so, slightly modifying the impedance.

If complex set impedance is required Zimp is a complex network, if a purely resistive set impedance is required Zimp is a resistor.

The value of the capacitor Cimp has to be high enough (advised value of 22 µF) in order to have an impedance negligible compare to Zimp or it may be used to generate the capacitive part of a complex impedance assuming that DC decoupling is kept.

3.1.3 Anti-sidetone network

Principle of operation

To avoid the microphone signal to come back with a too high level in the receive channel, the anti-sidetone circuit uses the microphone signal from pin SLPE (which is in opposite phase) to cancel the microphone signal at the IR input of the receive amplifiers. The anti-sidetone bridge principle already used for the TEA106x or the TEA111x families is used in a reversed way for the design of the anti-sidetone network as shown in fig. 13.

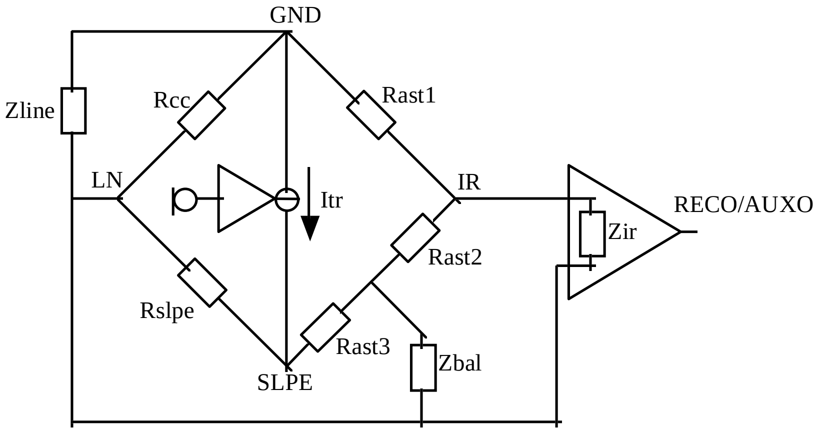

Fig. 13 Anti-sidetone bridge connection

This anti-sidetone bridge has the advantage of a relative flat transfer function in the audio frequency range between the line and the outputs RECO or AUXO, both with real and complex set impedances. Furthermore, the attenuation of the bridge for the receive signal (between pins LN and IR) is independent of the value chosen for Zbal after the set impedance has been fixed and the condition shown in equation (st) is fulfilled. Therefore, readjustment of the overall receive gain is not necessary in many cases.

The anti-sidetone circuit is composed of: Rcc//Zline, Rast1, Rast2, Rast3, Rslpe and Zbal. Maximum compensation is obtained when the following conditions are fulfilled:

$$\text{Rslpe} \times \text{Rast1} = \text{Rcc} \times (\text{Rast2} + \text{Rast3}) \qquad \text{(st)}$$

$$k = [\text{Rast2} \times (\text{Rast3} + \text{Rslpe})] / (\text{Rast1} \times \text{Rslpe})$$

$$\text{Zbal} = k \times \text{Zline}$$

The scale factor k is chosen to meet the compatibility with a standard value of capacitor for Zbal.

In practice, Zline varies strongly with line lenght and line type. Consequently, the value for Zbal has to be chosen to fit with an average line length giving acceptable sidetone suppression with short and long lines. The suppression further depends on the accuracy with which Zbal equals this average line impedance.

Example

Let’s optimize for a theorical equivalent average line impedance shown in Fig. 14.

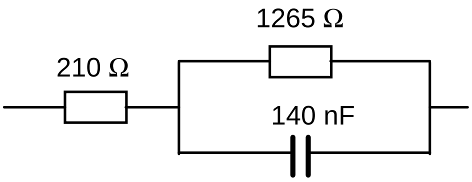

Fig. 14 Equivalent average line impedance

For compatibility of the capacitor value in Zbal with a standard capacitor value from the E6 series (220 nF):

k = 140 / 220 = **0.636**

For Rast2, a value of 3.92 kΩ has been chosen. So, using the previous equations, we can calculate Zbal, Rast1, Rast3. We find Rast1 = 130 kΩ, Rast3 = 390 Ω, and for Zbal **130 Ω in series with 220 nF // 820 Ω.**

The attenuation of the receive line signal between LN and IR can be derivated from the following equation:

Vir / Vln = (Zir // Rast2) / [Rast1 + (Zir // Rast2)]

If Rast2 >> (Rast3 // Zbal).

With the values used in this example, it gives 32 dB at 1 kHz.

Zir is the receive amplifier input impedance, typically 20 kΩ.

3.1.4 Automatic gain control

Principle of operation

The TEA1099 performs automatic line loss compensation. The automatic gain control varies the gains (according to the line DC current) of the amplifiers which send on line, except the DTMF one, and of the amplifiers which receive from line at pin IR. To enable this AGC function, the pin AGC must be connected to the pin LN. For line currents below a current threshold, Istart (typically 23mA), the gain control factor α is equal to 1, giving the maximum value to the gains. If this threshold current is exceeded, the gain control factor α is reduced and then the gains of the controlled amplifiers are also reduced. When the line current reaches an other threshold current, Istop (typically 57 mA), the gain control factor α is limited to its minimum value equal to 0.49 or 0.47, giving the lower value to the transmit and receive controlled gains. The gain control range of these amplifiers is typically 6.2 dB or 6.6 dB depending on the amplifier (see datasheet), which corresponds approximately to a line length of 5.5 km (0.5 mm twisted pair copper) with an attenuation of 1.2 dB/km.

The attenuation is correlated to the current Iagc sunk at pin AGC: when this current is lower than typically 4.8 µA the gains are maximum, when this current is higher than typically 12 µA the gains are minimum. This current is proportional to the voltage between pins SLPE and LN. There is an internal resistor which sets Istart and Istop, adding Ragc externally in series (between pins AGC and LN) reduces Iagc and increases the values of Istart and Istop.

Adjustments and performances

The AGC of the TEA1099 can be used with different exchange supply voltages and different feeding bridge resistances. For this purpose, a resistor Ragc, can be inserted between pins AGC and LN. This Ragc resistor increases the two threshold currents Istart and Istop. Fig.15 shows the control of the microphone gain versus the line current for two values of Ragc. When no AGC function is required, the AGC pin must be left open, then the control factor α equals to 1 and both controlled gains are at their maximum values.

When Ragc = 0 and the value of Istart is too high, increasing slightly the value of Rslpe reduces proportionally Istart and shifts the AGC to lower currents but the gains, the DC characteristic and the value of VBB are also modified. If the value of Rslpe has to be increased, it is possible to restore the typical gains (but not the value of VBB) by connecting in parallel an RC series network which makes a total AC impedance of 20 Ω.

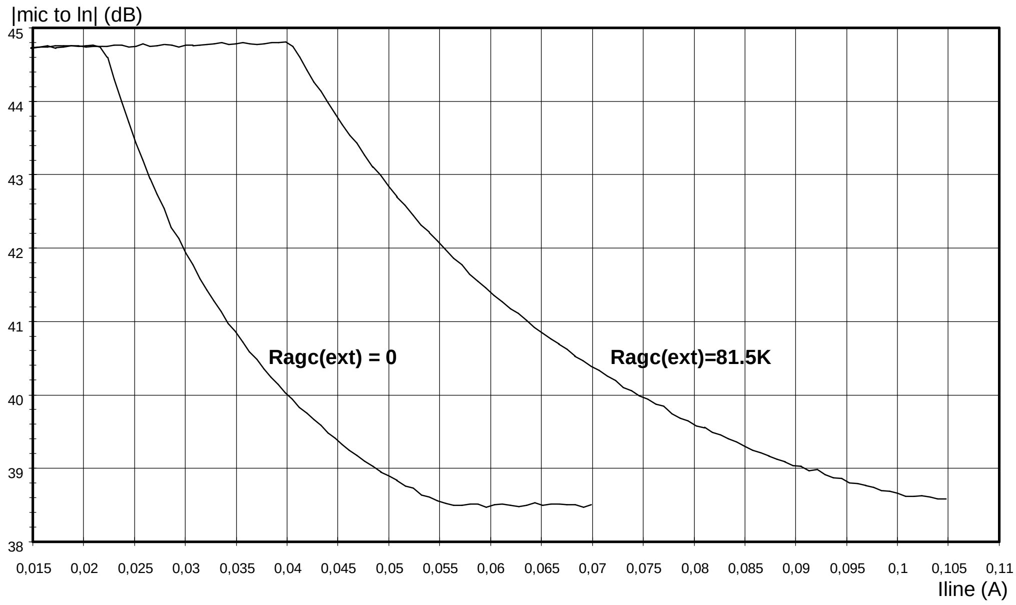

Fig. 15 AGC on the microphone gain versus line current and Ragc

3.2 Supplies

The TEA1099 provides three supply points, VBB is the strong supply for most of the internal circuitry and the amplifiers, VDD is a 3.35 V supply for the dialler or the microcontroller and MICS is a switched supply point for the electret microphones. Moreover, the TEA1099 can be powered by an external power supply connected at the input ESI.

Fig. 16 shows the block diagram of the supply block:

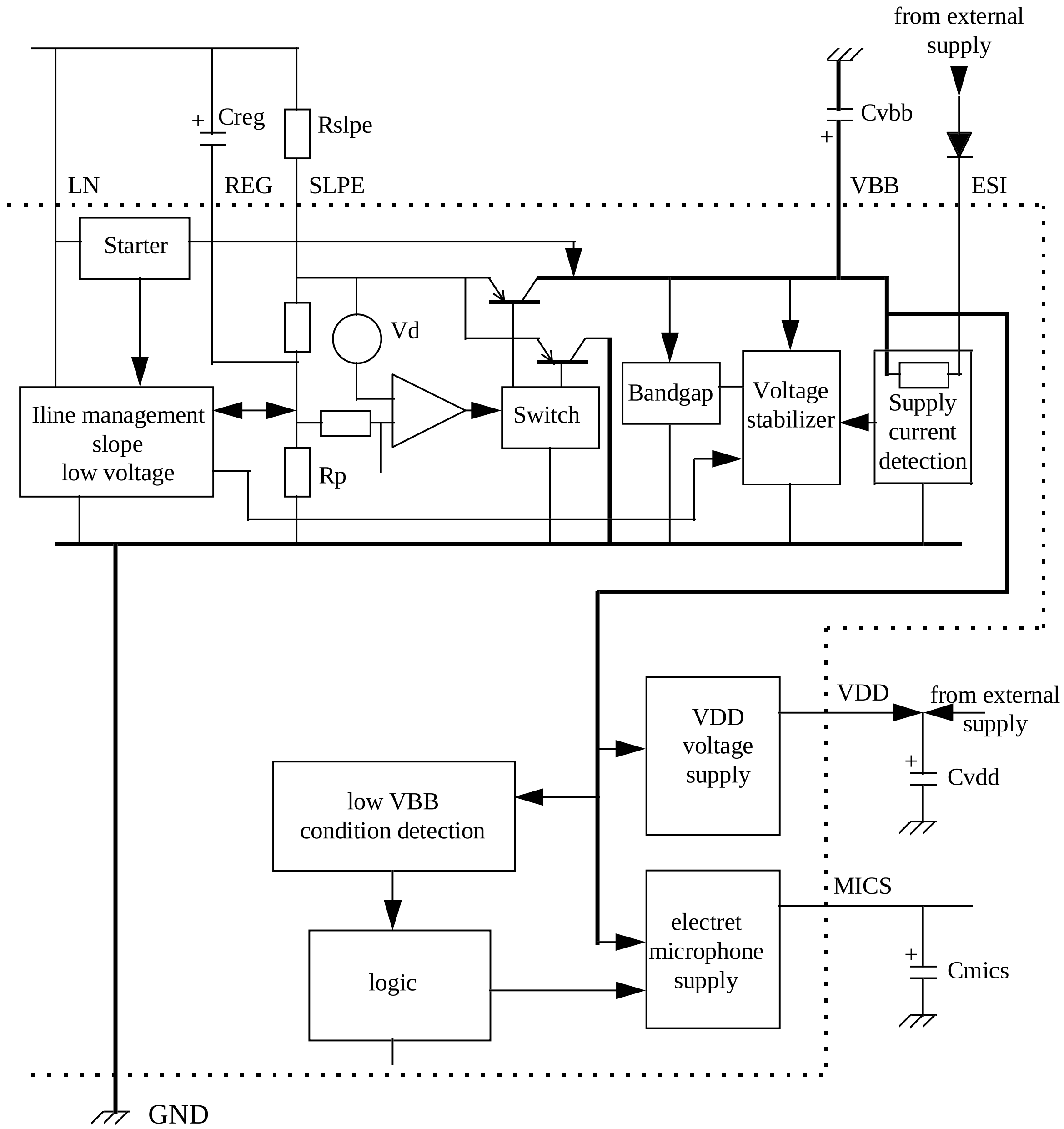

Fig. 16 Block diagram of the supply block

3.2.1 Supply VBB

Principle of operation

VBB can be either line powered or externally powered at ESI, when both supplies are available, the strongest of the two is automatically selected and used internally. When VBB is purely line powered, its value is correlated with the value of the line voltage and then of the line current as follows:

- below 9 mA : low voltage area

- 9 to 18 mA : VBB = 3 V
- 18 to 46 mA : VBB increases with a slope of 84 Ω from 3 to 5.35 V
- above 46 mA : VBB = 5.35 V

The correlation between line voltage and VBB is done in order to get a voltage difference between Vslpe and VBB of at least 0.7 V in order to guarantee a good power supply efficiency when AC signal is present on SLPE. On the block diagram, two PNP transistors drive the line current either to VBB or to GND: when the voltage on SLPE is higher than VBB + 0.3 V the current is driven to VBB, when the voltage on SLPE is lower than VBB the line current is driven to GND, when the voltage on SLPE is between VBB and VBB + 0.3 V both transistors are conducting in order to minimize distortion.

The correlation between line voltage and line current is done in order to get the optimized correlation between the power that can be extracted from the line and the power that can be delivered to the loudspeaker amplifier.

Fig. 7 shows these different voltages versus line current, fig. 17 shows the output voltage delivered on a 16 Ω and on a 50 Ω loudspeaker versus line current.

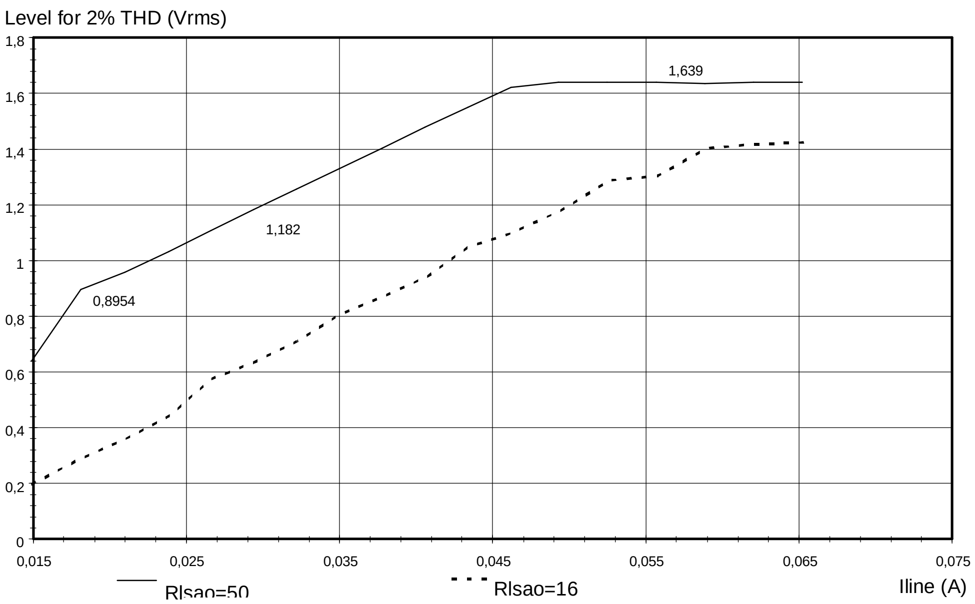

Fig. 17 Loudspeaker output power versus line current

The block diagram of Fig. 16 shows that an external power supply can be connected at ESI, the supply current detection block controls the selection of the supply used by sensing the current in a serial resistor. When the current is flowing from SLPE to VBB line current is used as power source, when current is flowing from ESI to VBB the external supply is then used and the shunt regulator built in the voltage stabilizer is adjusted to the external source (clamping VBB at 6.6 V) in order to get the value of VBB as close as possible to ESI value without extra current consumption. This shunt regulator is switched-off in power-down mode only and is still available in ringer mode.

A “low VBB condition detection” block detects if the value of VBB becomes higher than 2.9 V. The logic block is enabled when VBB becomes higher than 2.9 V. The handsfree and the loudspeaker amplifier parts are enabled when VBB becomes higher than 2.9 V.

When VBB becomes lower than 2.7 V, the VBB detector of the dynamic limiter on LSAO discharges the capacitor at pin DLC and the loudspeaker amplifier is disabled.

This block detects also if VBB becomes lower than 2.5 V. When VBB becomes lower than 2.5 V, the logic block is disabled and the handset speech mode is forced. Moreover, the handsfree and the loudspeaker amplifier are disabled.

The difference between 2.5 V and 2.9 V make hysteresis in order to keep stable behaviors.

VBB can be used to supply external circuits, in line powered condition the total amount of current drawn from VBB, MICS and from VDD must be low enough to stay compatible with the value of the line current.

A “starter” block is included in order to speed-up the charge of the capacitor Cvbb. This starter is active as soon as some voltage is available on the line when VBB is still lower than 2.4 V; when VBB is decreasing, it becomes active again when VBB becomes lower than 1.9 V.

Adjustments and performances

A capacitor Cvbb must be connected between pins VBB and GND, the advised value is 470 µF, a higher value would delay the start-up time of the system.

When an external voltage source is provided at ESI, a diode is necessary in order to allow VBB to take a value higher than the value of this source when the line current is high enough (e.g. 3.3 V at ESI and >46 mA of line current which provides VBB at 5.35 V) and to prevent VBB from collapse if this supply is temporarely out of order.

If an Rva resistor is connected between REG and GND to reduce the line voltage, the 0.7 V voltage difference between SLPE and VBB is reduced, then the power available for the loudspeaker amplifier is dramatically reduced and even can’t exist if the remaining voltage is lower than 0.3 V.

The “low VBB condition detector” enables the handsfree part , the loudspeaker amplifier and the logic block when VBB becomes higher than 2.9 V. The loudspeaker amplifier is disabled when VBB falls down below 2.7 V while the logic block is disabled when VBB falls down below 2.5 V. When VBB has fallen below 2.5 V, in order to indicate that the handset conversation mode is forced, the pin HFC is pulled to GND.

These hysteresis allow a stable operation of the loudspeaker amplifier in low supply condition because, if there is no change in the supply condition, the dynamic limiter should avoid VBB to fall below 2.7 V.

Fig. 18 shows the current consumption on VBB in ringer mode when no signal is sent to the loudspeaker amplifier and the input current on ESI necessary to get this value of VBB. The difference between these currents is available as power for the loudspeaker amplifier.

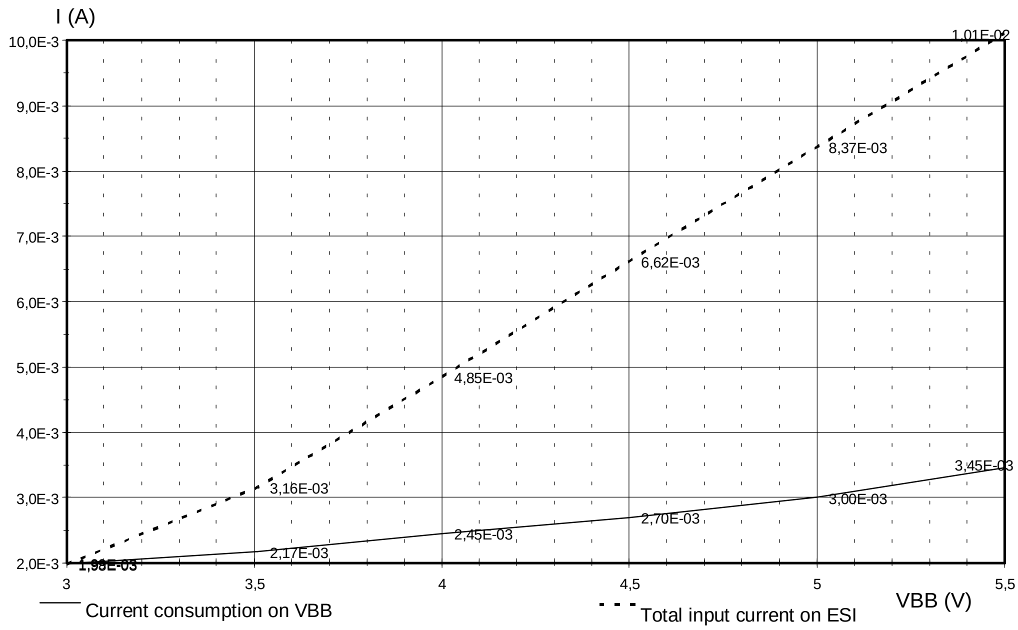

Fig. 18 Current consumption on VBB in "ring mode" versus VBB

3.2.2 Supply VDD

Principle of operation

The supply block VDD is fed from VBB, so VDD is typically 0.25 V lower than VBB and clamped typically at 3.35 V. Nevertheless the block VDD can be externally supplied, if the external source provides a current lower than 60 µA VDD is also clamped at 3.35 V, if this current is higher than 200 µA the voltage on VDD follows the voltage of the external source. These two modes allow either the supply of the dialler with a trickle current without any additional zener diode or the supply from an external regulated power supply without too much current consumption.

The output capability of VDD is typically 3 mA when ESI is supplied or when the line current is higher than 11 mA (with no extra consumption on VBB) in line powered mode. In line powered mode, this output capability is reduced progressively down to about 1 mA at 7 mA of line current depending on the current drawn at MICS.

In power down or ringer modes, VDD regulator is still working and provides 3.35 V as long as VBB is higher than 3.65 V. Moreover, the current consumption of VDD is lower than 150 nA when VDD ≤1.5 V in order to allow supply of the dialler or microcontroller with a trickle current.

Adjustments and performances

A capacitor Cvdd must be connected between pins VDD and GND even if this output is not functionaly used, in order to keep a small start-up time; its value must be limited to a few 10 µF. In power down mode (DC dialling or flash) the block VDD is still supplied from VBB and the capacitor Cvbb is the main tank.

When an external voltage source is provided, a diode in series is mandatory if this source is lower than 3.5 V or if it may be switched-off. If it is higher than 3.5 V the current consumption is less than 200 µA.

Fig. 19 shows the typical current consumption on VDD in trickle mode when VBB = O.

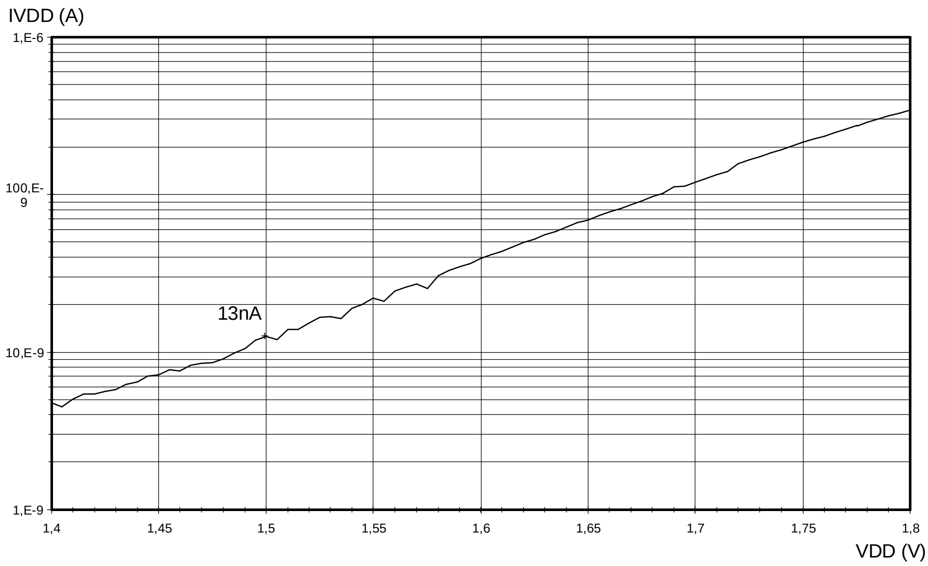

Fig. 19 Current consumption on VDD at VBB = 0

3.2.3 Microphone supply MICS

Principle of operation

The electret microphone supply block is fed from VBB and provides a 2 V regulated supply with a capability of 1 mA. The output impedance is typically 200 Ω and must be filtered with a capacitor referenced to GNDTX. This output is switched-off in power down mode and in modes where the electret microphones are not necessary (see list in fig. 4).

Adjustments and performances

When the output is filtered with a 10 µF capacitor to GNDTX, the noise at MICS is typically -114 dBVp. The value of this capacitor fixes the impedance of this supply point; when the value of this capacitor is too small, the attenuation of the handset microphone signal may not be sufficient compare to the switching range of the duplex controller.

3.3 Transmit

The selection of the signal transmitted on line is made according to the table of fig. 4. This signal comes from the four following amplifiers: handset microphone amplifier, DTMF amplifier, handsfree microphone channel and auxiliary transmit amplifier.

Fig. 20 shows the block diagram of the transmit part.

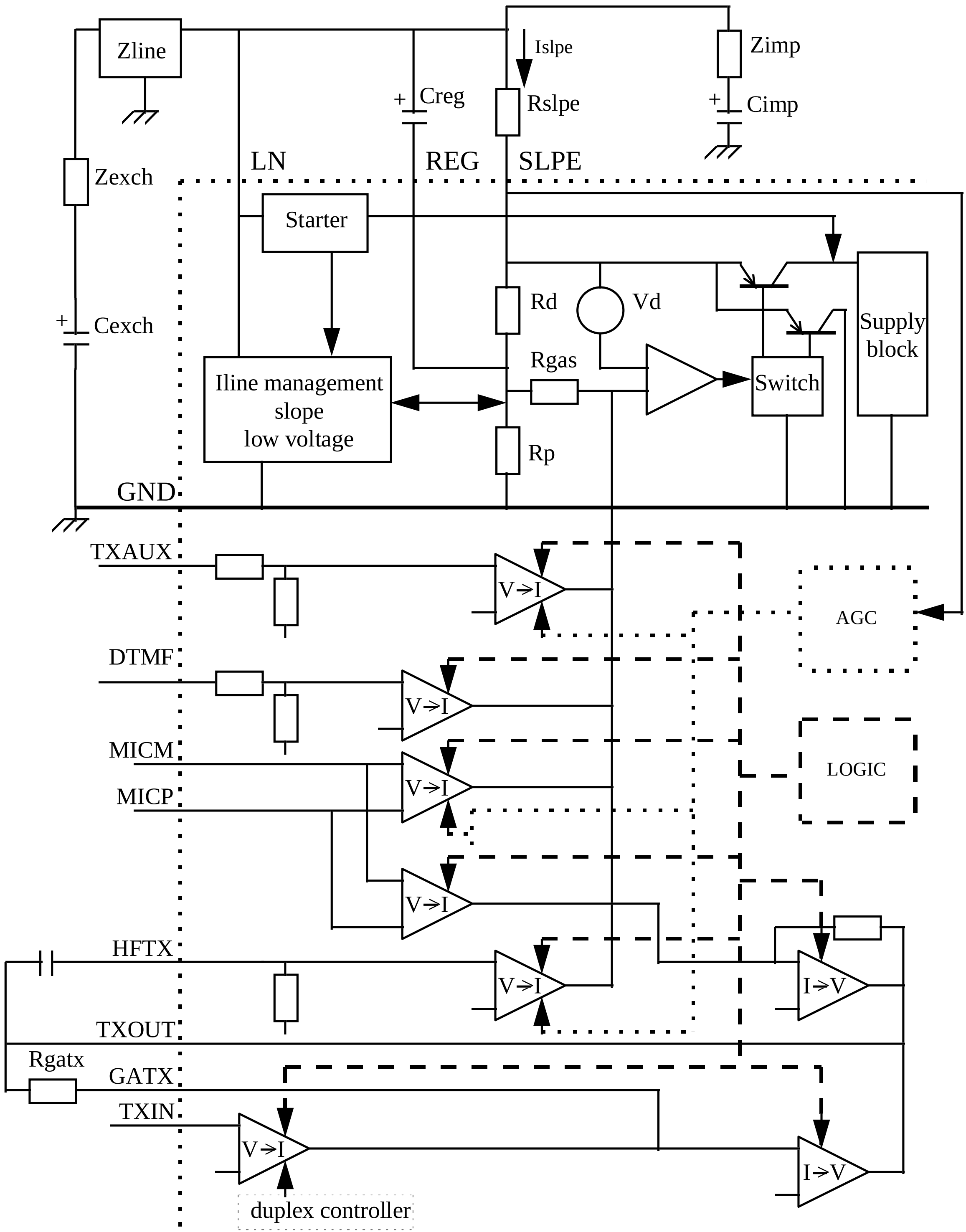

Fig. 20 Block diagram of the transmit part

3.3.1 Handset microphone amplifier

Principle of operation

The microphone amplifier has symmetrical high input impedances (typically 70 kΩ -2 times 35 kΩ- between pins MIC+ and MIC- with maximum tolerances of +/- 15%).

As can be seen from fig. 20, the microphone amplifier to LN is built up out of two parts: a preamplifier which realizes a voltage to current conversion, and an end-amplifier (common with the other three transmit paths) which realizes the current to voltage conversion. The overall gain Gv(mic-ln) of the microphone amplifier from inputs MIC+/MIC- to output LN is internally set and given by the following equation:

$$Gv(mic\text{-}ln) = 20 \times \log Avmic$$

$$Avmic = 5.7 \times (Rgasint / Rrefint) \times (Ri//Zl / Rslpe) \times \alpha$$

with:

Ri = the AC apparatus impedance, Rcc//Rp (typically 620 Ω // 25 kΩ)

Rgasint = internal resistor realizing the current to voltage conversion (typically 27.6 kΩ with a spread of +/-15%)

Rrefint = internal resistor determining the current of an internal current stabilizer (typically 14.7 kΩ with a spread of +/- 15% correlated to the spread of Rgasint)

Zl = load impedance of the line during the measurement

α = gain control factor varying from 1 at Iline = 15 mA to 0.49 at Iline = 70 mA when AGC function is applied (see chapter 3.1.4 for details)

Using these typical values in the equation and assuming Zline = 600 Ω, we find a gain equal to:

$$Gv(mic\text{-}ln) = 20 \times \log Avmic = 44.1\ dB \quad \text{at Iline} = 15\ mA$$

The AGC gain control acts on the microphone preamplifier stage, modifying its transconductance. Moreover the logic block enables or disables the preamplifier according to the selected mode (see fig. 4).

The inputs are biased at one Vd. The input of this microphone amplifier is able to handle AC signals up 18 mVrms with less than 2% total harmonic distortion.

The microphone amplifier to TXOUT is used for monitoring the microphone signal in order to built an external antihowling circuit (§3.4.5). It is also built up out of two parts: a preamplifier which realizes a voltage to current conversion, and an end-amplifier (common with the handsfree microphone amplifier) which realizes the current to voltage conversion. The overall gain Gv(mic-txout) of the microphone amplifier from inputs MIC+/MIC- to output TXOUT is fixed and given by the following equation:

$$Gv(mic\text{-}txout) = 20 * \log (6.4 * Rgatxint / Rrint).$$

Rgatxint = internal resistor (typically 80 kΩ with a spread of +/-15%)

Rrint = internal resistor (typically 1.66 kΩ with a spread of +/-15%)

Adjustments and performances

Fig. 21 shows the typical frequency response of the microphone amplifier of the TEA1099.

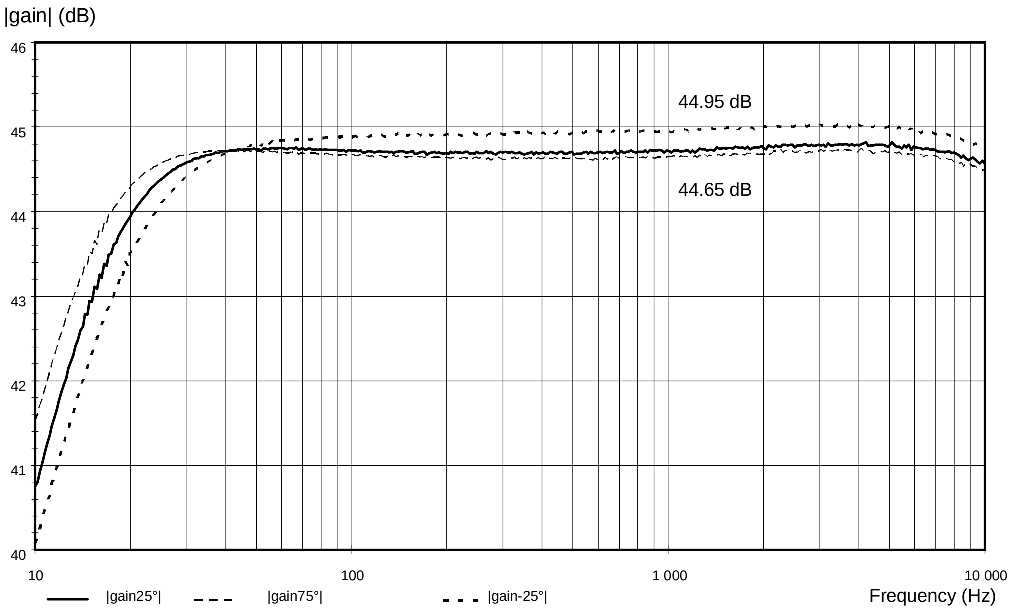

Fig. 21 Handset microphone gain versus frequency: influence of temperature

Fig. 22 shows the distortion of the signal on the line as a function of the microphone signal at nominal DC settings and for a different line currents.

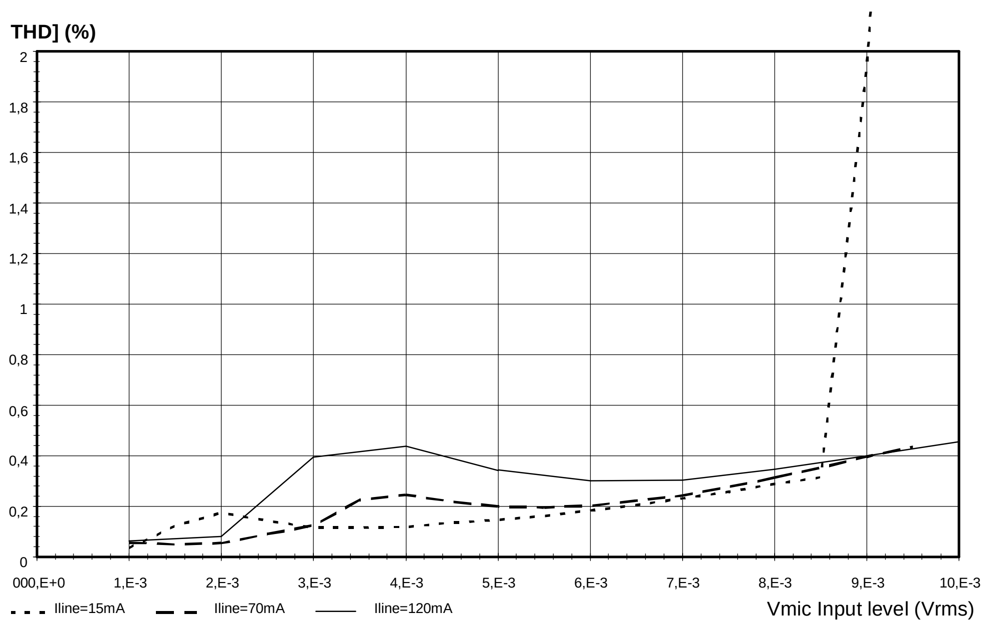

Fig. 22 Distortion on line versus handset microphone signal on TEA1099

Fig. 23 shows the distortion of the line signal versus level at line current of 4 mA.

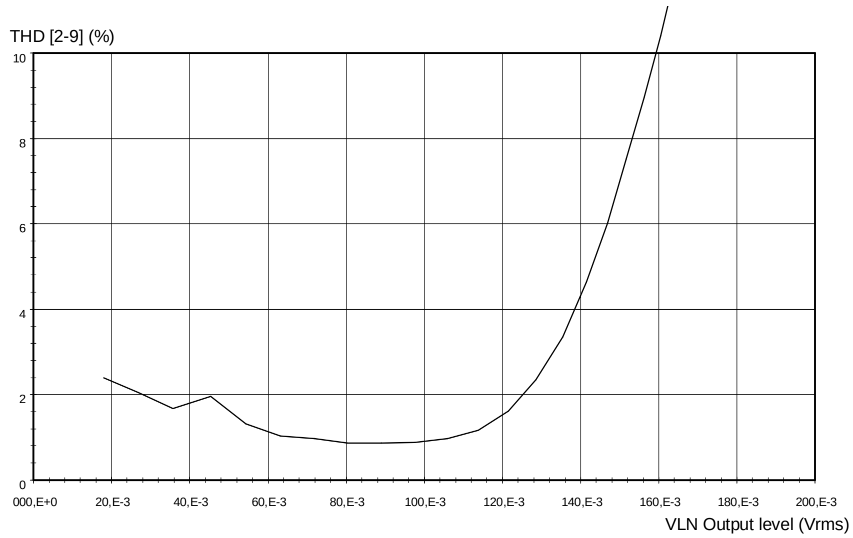

Fig. 23 Distortion of line signal at Iline = 4 mA

Fig. 24 shows the microphone noise (psophometrically weighted: P53 curve) versus line current when a 200 Ω resistor is connected between the inputs MIC- and MIC+.

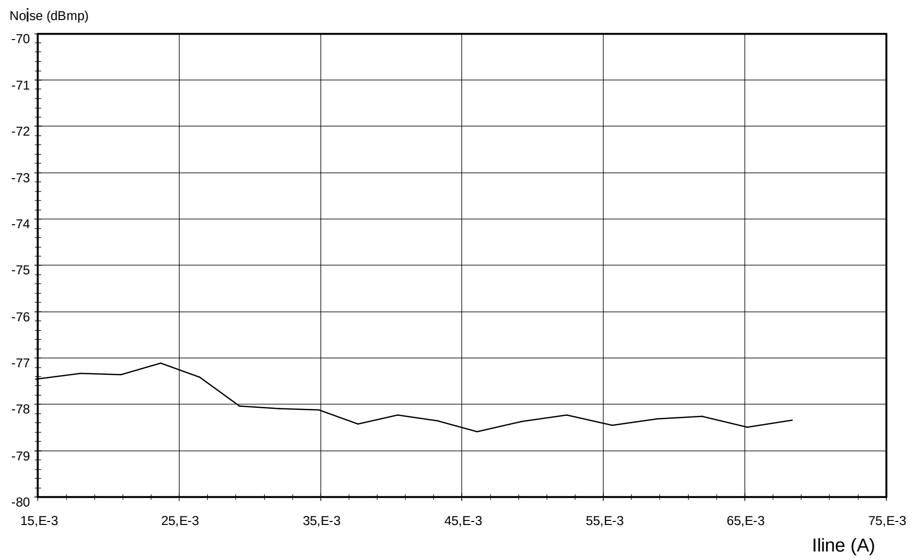

Fig. 24 Handset microphone noise versus line current

Fig. 25 shows the common mode rejection ratio at 15 mA. Two curves are present in this fig. 25, the first one is the spectrum of the signal on pin LN when a microphone signal is applied on pin MIC- while pin MIC+ is shorted to GND, the second one is the spectrum of the signal on pin LN when a microphone signal is applied on pins

MIC- and MIC+ shorted together. Both signals are at 1 kHz, the difference between the two curves gives the CMRR.

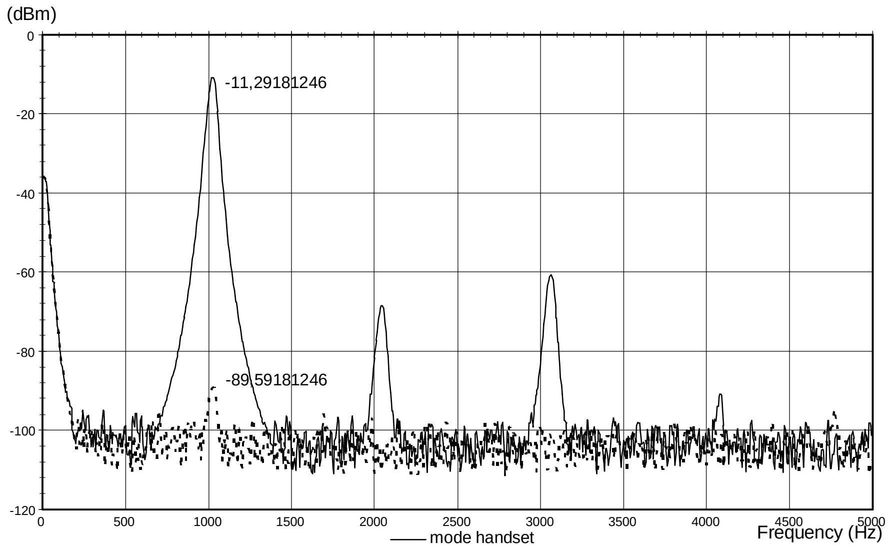

Fig. 25 Common mode rejection ratio on microphone

3.3.2 DTMF amplifier

Principle of operation

The DTMF amplifier has an a-symmetrical high input impedance of 20 kΩ between pins DTMF and GND with a maximum spread of +/-15%. The input is biased at GND, so if the input signal is polarized at GND there is no need of decoupling capacitor in series. The DTMF amplifier is built up out of three parts: an attenuator by a factor of 7.15, a preamplifier which realizes the voltage to current conversion and the same end-amplifier as the handset microphone amplifier. No AGC is applied to the DTMF channel. The overall gain Gv(dtmf-ln) of the DTMF amplifier from input DTMF to output LN is given by the following equation:

$$Gv(dtmf\text{-}ln) = 20 \times \log Avmf$$

$$Avmf = 0.66 \times (Rgasint / Rrefint) \times (Ri//Zl / Rslpe)$$

with:

Ri = the AC apparatus impedance, Rcc//Rp (typically 620 Ω // 25 kΩ)

Rgasint = internal resistor realizing the current to voltage conversion (typically 27.6 kΩ with a spread of +/-15%)

Rrefint = internal resistor determining the current of an internal current stabilizer (typically 14.7 kΩ with a spread of +/- 15% correlated to the spread of Rgasint)

Zl = load impedance of the line during the measurement

Using these typical values in the equation and assuming Zline = 600 Ω, we find a gain equal to:

$$Gv(dtmf\text{-}ln) = 20 \times \log Avmf = 25.4 \text{ dB}$$

Fig. 26 shows the frequency response of the DTMF amplifier at 15 mA and different temperatures.

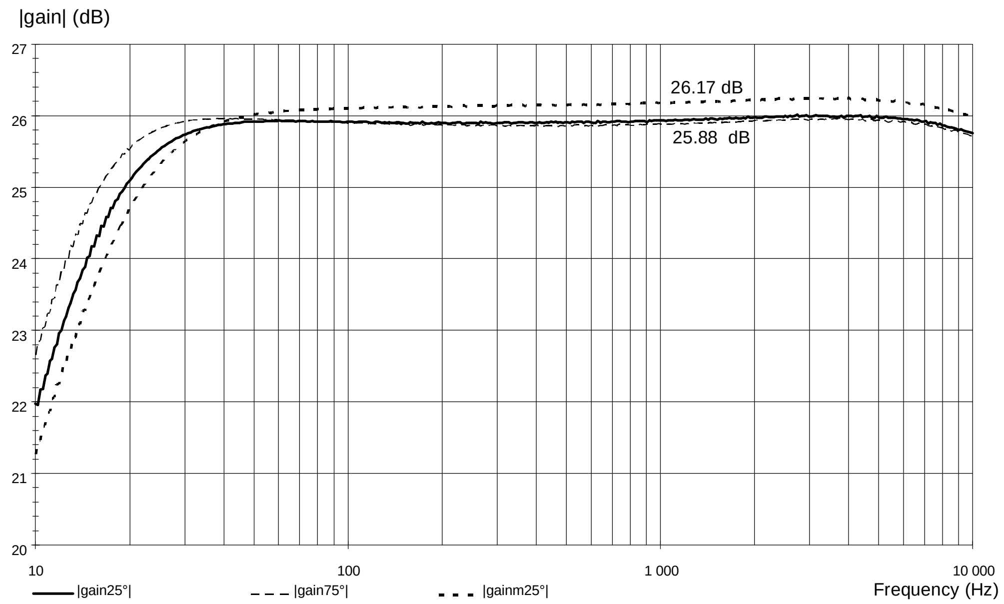

Fig. 26 DTMF gain versus frequency: influence of the temperature

The input of the DTMF amplifier can handle signals up to 180 mVrms with less than 2% THD. Fig. 27 shows the distortion on line versus the rms input signal at different line currents.

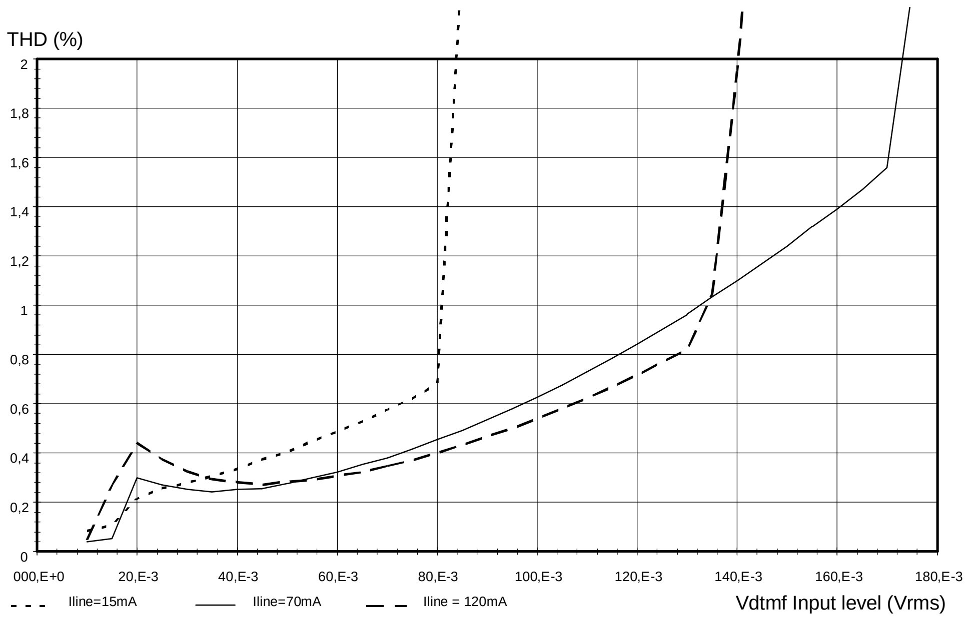

Fig. 27 Distortion of the DTMF signal on line versus input signal

3.3.3 Handsfree microphone channel

Principle of operation

Fig. 20 shows that the handsfree microphone channel is splitted into two blocks: the handsfree microphone amplifier (TXIN to TXOUT) and the HFTX amplifier.

The handsfree microphone amplifier is referenced to pin GNDTX. This is in order to prevent interference from other blocks of the TEA1099 or of the application, GNDTX is called a clean ground. The input and output signals of the microphone channel have to be referenced to GNDTX. Pin GNDTX itself has to be shorted to GND.

The input of the handsfree microphone amplifier is pin TXIN. It is an a-symetrical input well suited for electret microphones. Induced signals in the short wire between the microphone and pin TXIN are assumed to be negligible. This is in contrary with the handset microphone which is connected via the handset cord.

The output of the microphone amplifier is pin TXOUT. In handsfree mode, pin TXOUT has to be connected via a decoupling capacitor to pin HFTX.

As can be seen in fig. 20, between pins TXIN and TXOUT, this microphone amplifier is built up out of two parts: a preamplifier and an end-amplifier. The gain of the preamplifier is determined by the duplex controller block, see § 3.5. The gain of the end-amplifier is determined by the external feedback resistor Rgatx.

The overall gain Gv(txin-txout) of the microphone amplifier from input TXIN to output TXOUT in TX-mode is given as:

$$Gv(txin\text{-}txout) = 20 * \log (0.7 * Rgatx / Rstab).$$

With Rstab being the resistor at pin STAB of 3.65 kΩ.

The HFTX amplifier has an a-symmetrical high input impedance of 20 kΩ between pins HFTX and GND with a maximum spread of +/-15%. The HFTX amplifier is built up out of two parts: a preamplifier which realizes the voltage to current conversion and the same end-amplifier as the handset microphone amplifier. The overall gain Gv(hftx-ln) of the HFTX amplifier from input HFTX to output LN is given by the following equation:

$$Gv(hftx\text{-}ln) = 20 \times \log Avhft$$

$$Avhft = 1.93 \times (Rgasint / Rrefint) \times (Ri//Zl / Rslpe) \times \alpha$$

with:

Ri = the AC apparatus impedance, Rcc//Rp (typically 620 Ω // 25 kΩ)

Rgasint = internal resistor realizing the current to voltage conversion (typically 27.6 kΩ with a spread of +/-15%)

Rrefint = internal resistor determining the current of an internal current stabilizer (typically 14.7 kΩ with a spread of +/- 15% correlated to the spread of Rgasint)

Zl = load impedance of the line during the measurement

α = gain control factor varying from 1 at Iline = 15 mA to 0.47 at Iline = 70 mA when AGC function is applied (see chapter 3.1.4 for details)

Using these typical values in the equation and assuming Zline = 600 Ω, we find a gain equal to:

$$Gv(hftx\text{-}ln) = 20 \times \log Avhft = 34.7 \text{ dB} \quad \text{at } Iline = 15 \text{ mA}$$

Adjustments and performances

A handsfree microphone, referenced to GNDTX, can be connected to the input TXIN via a DC blocking capacitor Ctxi. Together with the input impedance at pin TXIN of 20 kΩ, this capacitor form a first order high-pass filter which can be used to adjust the transmit curve.

The handsfree electret microphone can be supplied from MICS via a resistor.

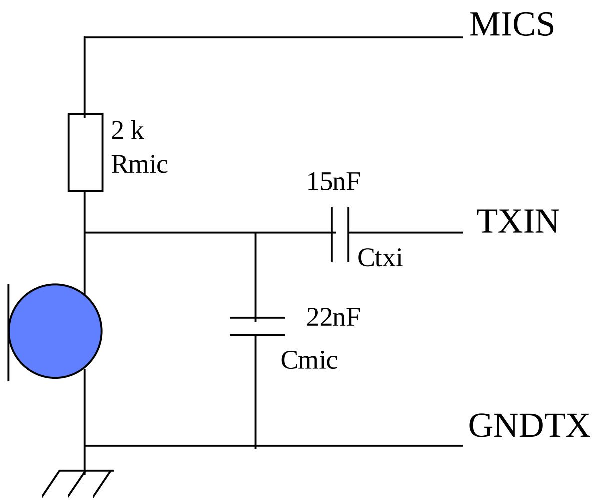

Fig. 28 Connection of the handsfree electret microphone

The sensitivity of the electret microphone is set via resistor Rmic. By putting a capacitor Cmic in parallel with the microphone, a first order low-pass filter is formed for the microphone signal in order to adjust the transmit curve.

Via the resistor Rgatx, the gain of the microphone amplifier can be adjusted from +6 to +31 dB to suit application specific requirements. With the resistor Rgatx = 30 kΩ, the gain equals typically 15 dB.

Capacitor Cgatx can be applied in parallel with resistor Rgatx to provide a first order low-pass filter for the adjustment of the transmit curve.

The input of the microphone amplifier can handle signals up to 18 mVrms with 2% total harmonic distortion. However, the microphone input signal is also used by the duplex controller. At 7 mVrms at the input, the positive part of the signal on pin TSEN starts clipping which might influence the switching behavior. It is therefore advisable to keep the microphone input signal below this level.

The TXOUT output of the TEA1099 has an internal impedance of 200 Ω and its output drive capability is 20 µArms.

The output noise at TXOUT of the TEA1099 is -103 dBVp (psophometrically weighted) at a gain of 15 dB. With the sending gain between HFTX and LN set at 35 dB (total handsfree transmit gain of 50 dB), the noise level on the line will be : -66 dBmp.

In Tx-mode, the noise level will be at its maximum. In Ix-mode and Rx-mode, the noise at TXOUT will be lower because the contribution of the preamplifier is reduced. However, the bottom level of the sending noise at TXOUT is limited by the end-amplifier and is about -111 dBVp. So, in Ix-mode, the noise level on line will be: -74 dBmp.

The input of the HFTX amplifier is biased at 2 Vd and can handle signals up to 95 mVrms to LN with less than 2% THD. Fig. 29 shows the distortion on line versus the rms input signal at Iline = 70 mA.

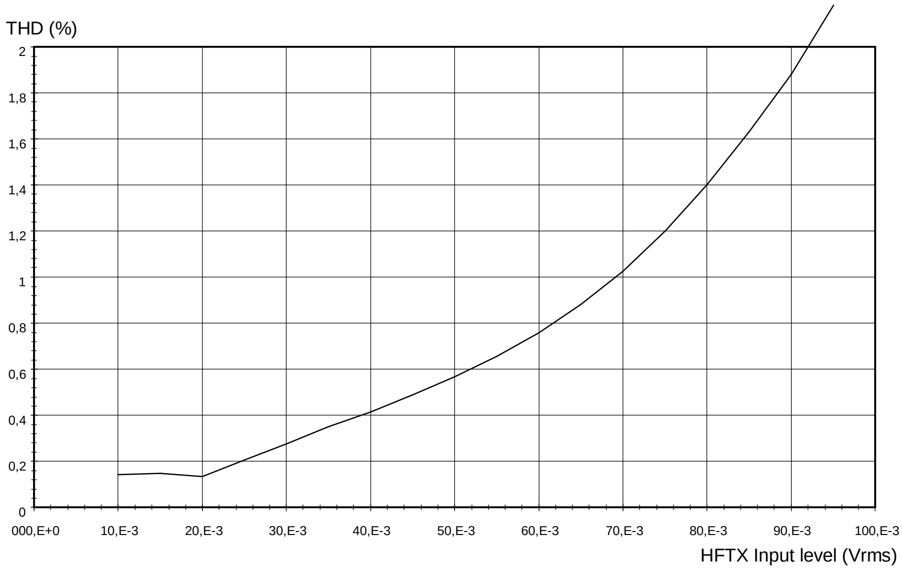

Fig. 29 Distortion on line versus HFTX input level

3.3.4 Auxiliary transmit amplifier TXAUX

Principle of operation

The auxiliary transmit amplifier has an a-symmetrical high input impedance of 20 kΩ between pins TXAUX and GND with a maximum spread of +/-15%. The auxiliary transmit amplifier is built-up out of two parts: a preamplifier which realizes the voltage to current conversion and the same end-amplifier as the handset microphone amplifier. The overall gain Gv(txaux-ln) of the auxiliary transmit amplifier from input TXAUX to output LN is given by the following equation:

$$Gv(txaux\text{-}ln) = 20 \times \log Avtxa$$

$$Avtxa = 0.151 \times (Rgasint / Rrefint) \times (Ri//Zl / Rslpe) \times \alpha$$

with:

Ri = the AC apparatus impedance, Rcc//Rp (typically 620 Ω // 25 kΩ)

Rgasint = internal resistor realizing the current to voltage conversion (typically 27.6 kΩ with a spread of +/-15%)

Rrefint = internal resistor determining the current of an internal current stabilizer (typically 14.7 kΩ with a spread of +/- 15% correlated to the spread of Rgasint)

Zl = load impedance of the line during the measurement

α = gain control factor varying from 1 at Iline = 15 mA to 0.47 at Iline = 70 mA when AGC function is applied (see chapter 3.1.4 for details)

Using these typical values in the equation and assuming Zline = 600 Ω, we find a gain equal to:

$$Gv(txaux\text{-}ln) = 20 \times \log Avtxa = 12.6\ dB \text{ at } Iline = 15\ mA$$

The input of the TXAUX auxiliary amplifier is biased at two Vd and can handle signals up to 1 Vrms with less than 2% THD and signals up to 50 mVrms with less than 0.1 % THD. Fig. 30 shows the distortion on line versus the rms input signal at Iline = 70 mA.

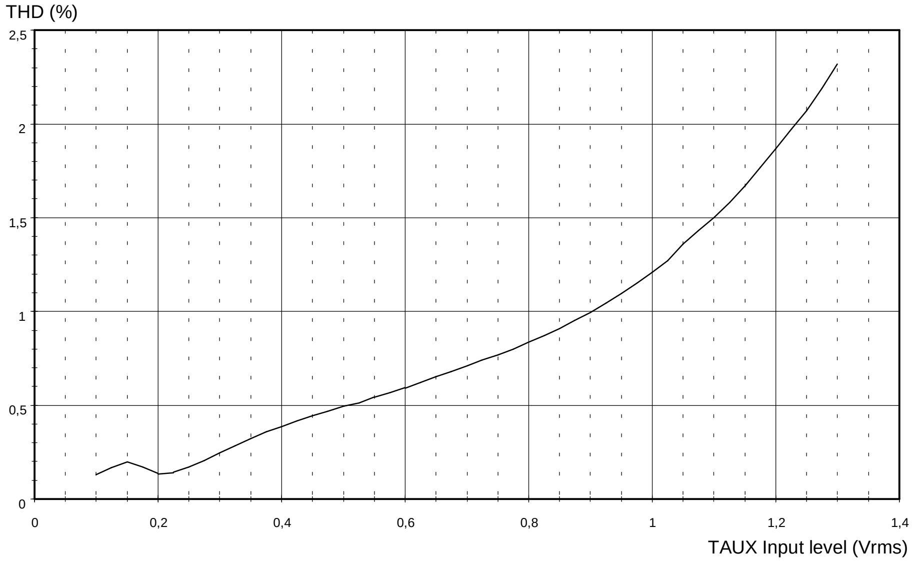

Fig. 30 Distortion on line versus TXAUX input signal

Fig. 31 shows the transmit noise (psophometrically weighted: P53 curve) versus line current when a 2 kΩ resistor is connected between the inputs TXAUX and GND.

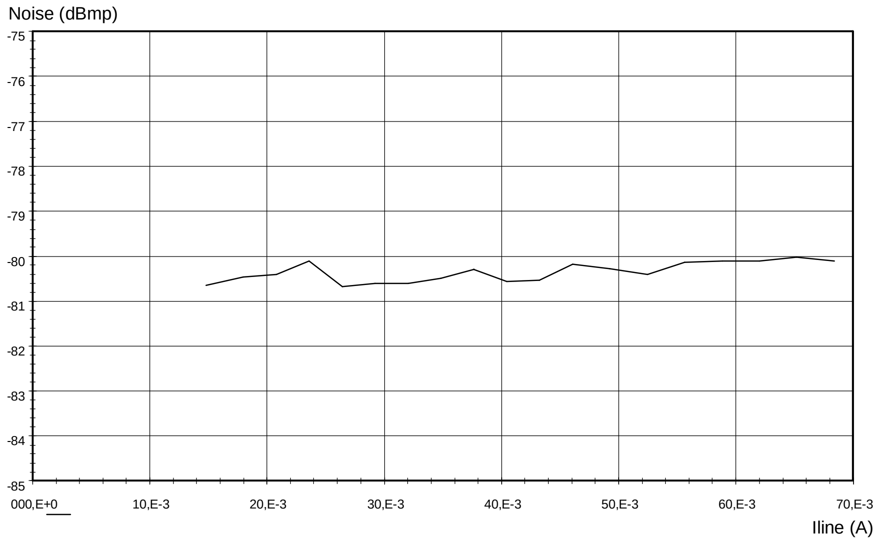

Fig. 31 Transmit noise versus line current

3.4 Receive

The receive part includes four different amplifier outputs: line receive amplifier RECO, earphone amplifier QR, loudspeaker amplifier LSAO, auxiliary receive amplifier AUXO and two different inputs: IR from line and RAUX from auxiliary. The selection of the receive signal is made according to the table of fig. 4.

Fig. 32 shows the block diagram of the receive part while fig. 33 shows the block diagram related to the auxiliary output AUXO.

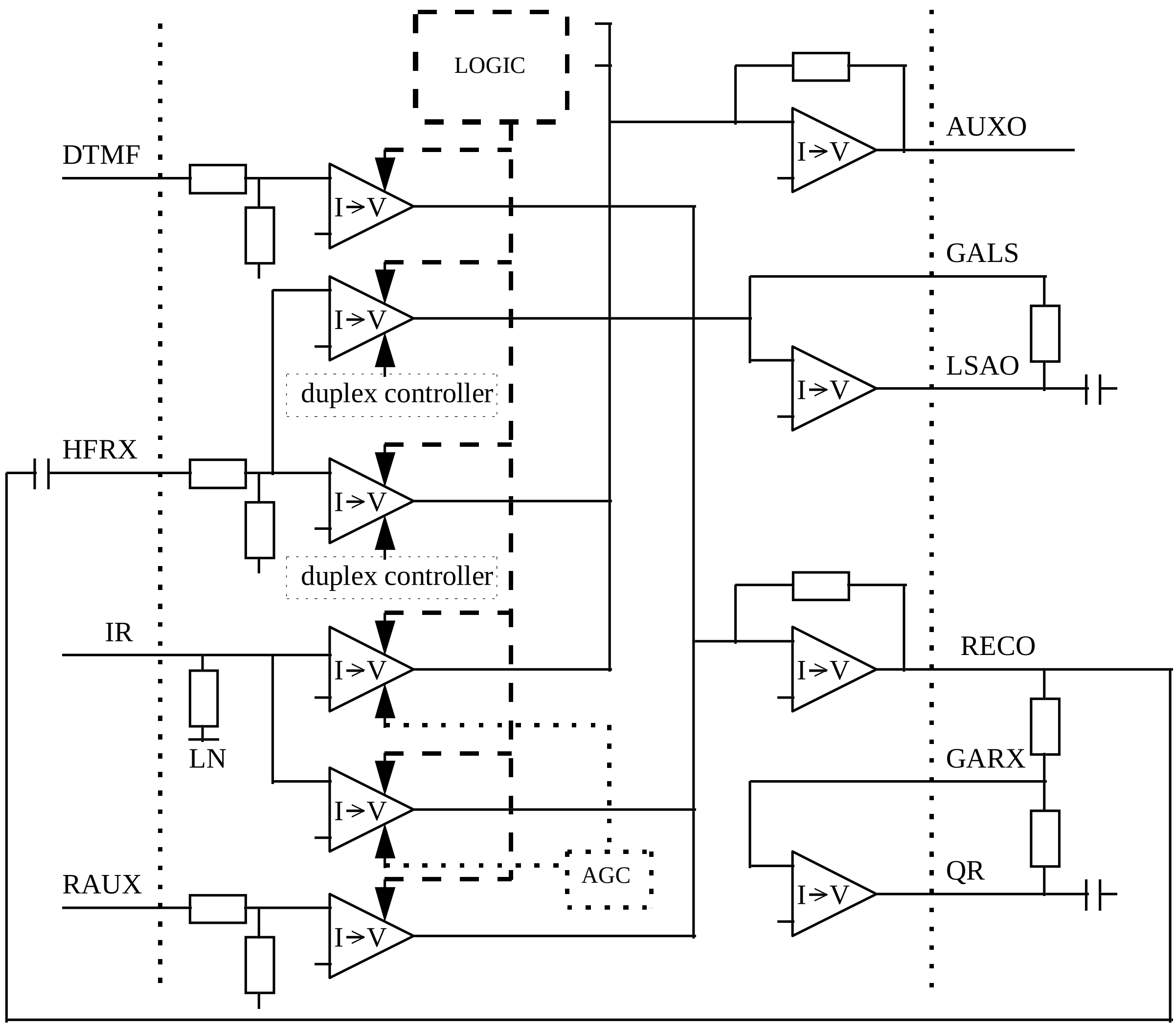

Fig. 32 Receive block diagram

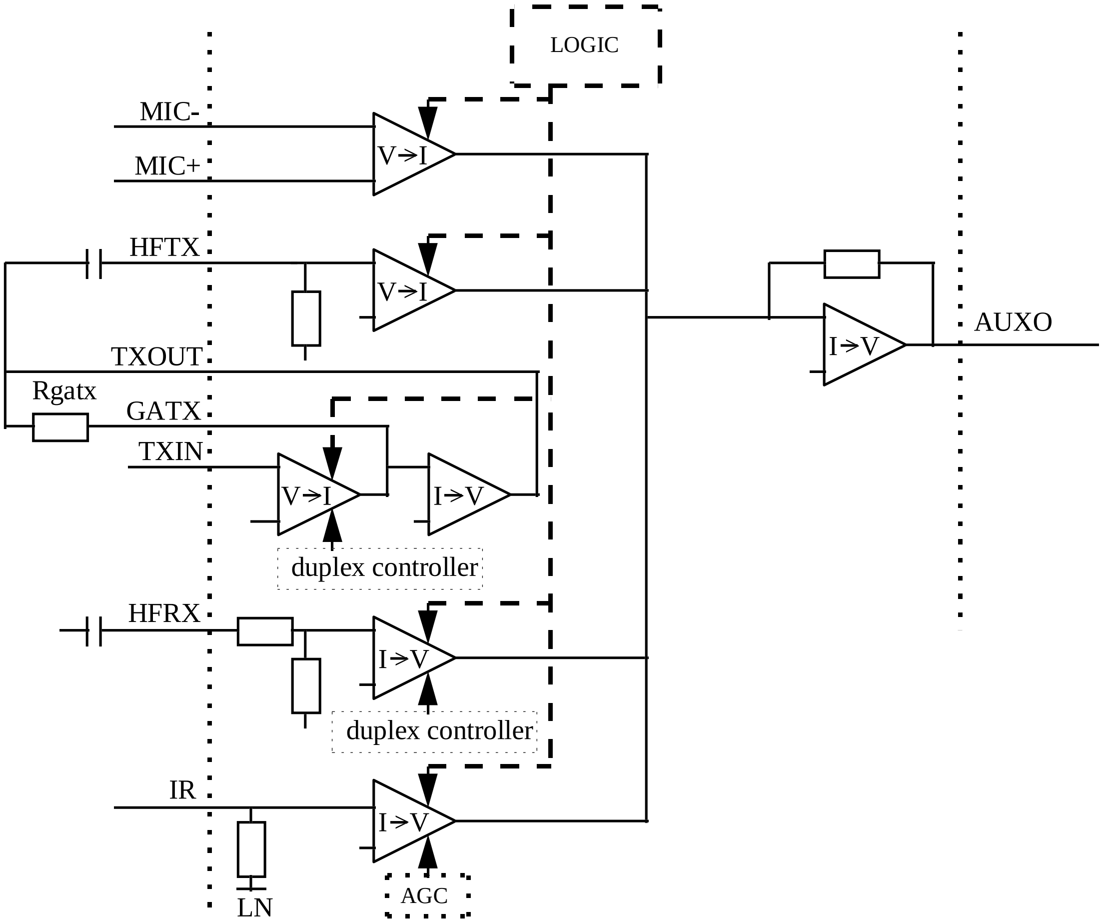

Fig. 33 Block diagram related to AUXO

3.4.1 Line receive amplifier RECO

Principle of operation

According to the logic selection (see fig. 4), the line receive amplifier can get signal from three different inputs: IR for line signals, DTMF for confidence tone and RAUX as auxiliary input.

As can be seen from fig. 32, the line receive amplifier itself is built up out of four parts: three preamplifiers (inputs IR, DTMF and RAUX) which realize a voltage to current conversion and an end-amplifier which realizes the current to voltage conversion.

The RECO output of the TEA1099 has an internal impedance of 125 Ω and is able to drive loads down to an impedance of 5 kΩ.

IR has an a-symmetrical high input impedance between pins IR and LN. It is equal to 20 kΩ with a maximum tolerance of +/-15%. The overall gain Gv(ir-reco) of the receive amplifier from input IR to output RECO is given by the equation:

$$Gv(ir\text{-}reco) = 20 \times \log Avrx$$

$$Avrx = 3.5 \times Rgarint/Rrefint \times \alpha$$

with:

Rgarint = internal resistor realizing the current to voltage conversion (typically 128 kΩ with a spread of +/-15%)

Rrefint = internal resistor determining the current of an internal current stabilizer (typically 14.7 kΩ with a spread of +/- 15% correlated to the spread of Rgarint)

α = gain control factor varying from 1 at Iline = 15 mA to 0.49 at Iline = 70 mA when AGC function is applied (see chapter 3.1.4 for details)

Using these typical values in the equation, we find a gain equal to:

Gv(ir-reco) = 20 $\times$ log Avrx = 29.7 dB   at Iline = 15 mA

DTMF has an a-symmetrical high input impedance between pins DTMF and GND shared with the DTMF amplifier. It is equal to 20 kΩ with a maximum tolerance of +/-15%. The overall gain Gv(dtmf-reco) of the receive amplifier from input DTMF to output RECO is given by the equation:

Gv(dtmf-reco) = 20 $\times$ log Avmfe

Avmfe = 0.017 $\times$ Rgarint/Rrefint

This gain is not affected by the AGC, using these typical values in the equation, we find a gain equal to:

Gv(dtmf-reco) = 20 $\times$ log Avmfe = -16.5 dB   at Iline = 15 mA

RAUX has an a-symmetrical high input impedance between pins RAUX and GND. It is equal to 20 kΩ with a maximum tolerance of +/-15%. The overall gain Gv(raux-reco) of the receive amplifier from input RAUX to output RECO is given by the equation:

Gv(raux-reco) = 20 $\times$ log Avrrax

Avrrax =0.088 $\times$ Rgarint/Rrefint

This gain is not affected by the AGC, using these typical values in the equation, we find a gain equal to:

Gv(raux-reco) = 20 $\times$ log Avrrax = -2.3 dB   at Iline = 15 mA

Adjustments and performances

29.7 dB of receive gain between IR and RECO compensate approximately the attenuation provided by the antisidetone network minus 2 dB.

Fig. 34 shows the frequency response of the line receive amplifier from IR to RECO at different temperatures.

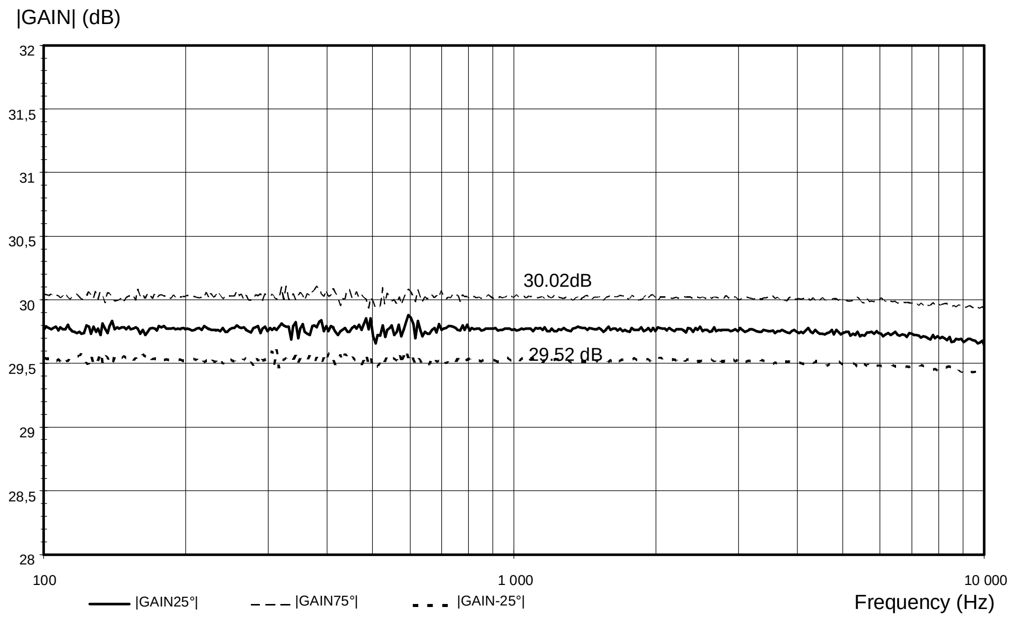

Fig. 34 Receive gain versus frequency: influence of temperature

The output is biased at 2 Vd with a temperature drift of -4 mV/°C, so the maximum output swing on RECO depends hardly on the value of VBB.

The receive input IR can handle signals up to 50 mVrms with less than 2% THD. Fig. 35 shows the distortion on RECO when the limitation is related to the input voltage at IR for a line current equal to 70 mA.

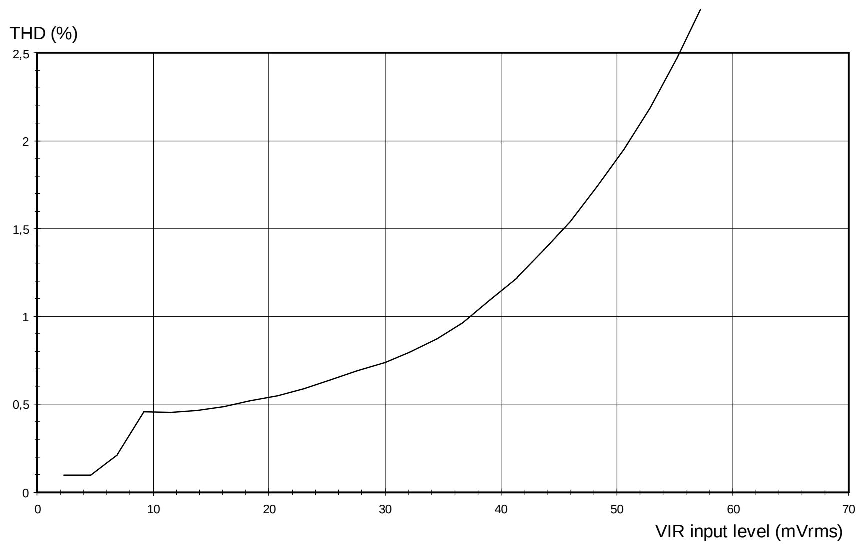

Fig. 35 Distortion on RECO versus input signal on IR

The receive input RAUX is biased at 2 Vd and can handle signals up to 900 mVrms with less than 2% THD. Fig. 36 shows the distortion on RECO when the limitation is related to the input voltage at RAUX for a line current equal to 15 mA.

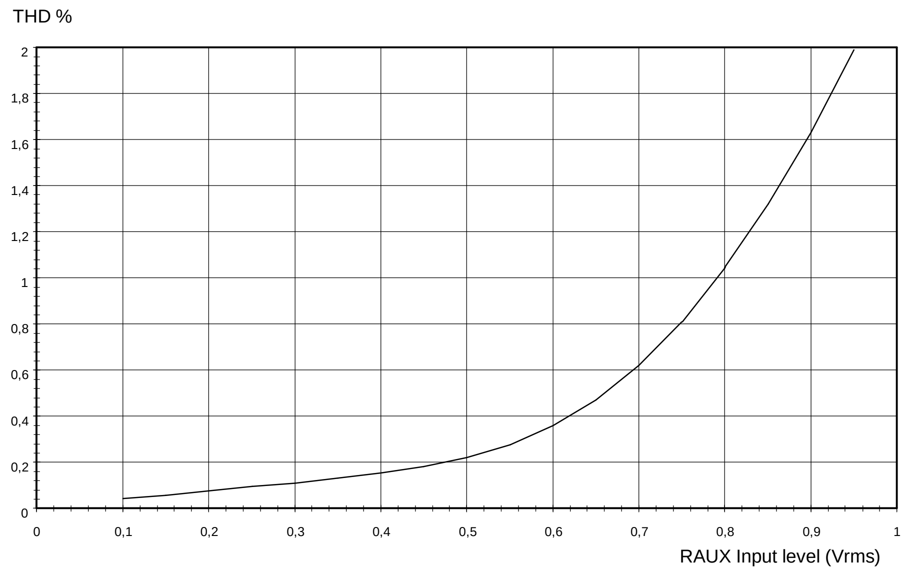

Fig. 36 Distortion on RECO versus input signal on RAUX

Fig. 37 shows the distortion of the signal on RECO as a function of the rms signal on RECO with a load of 5 kΩ and a line current of 15 mA.

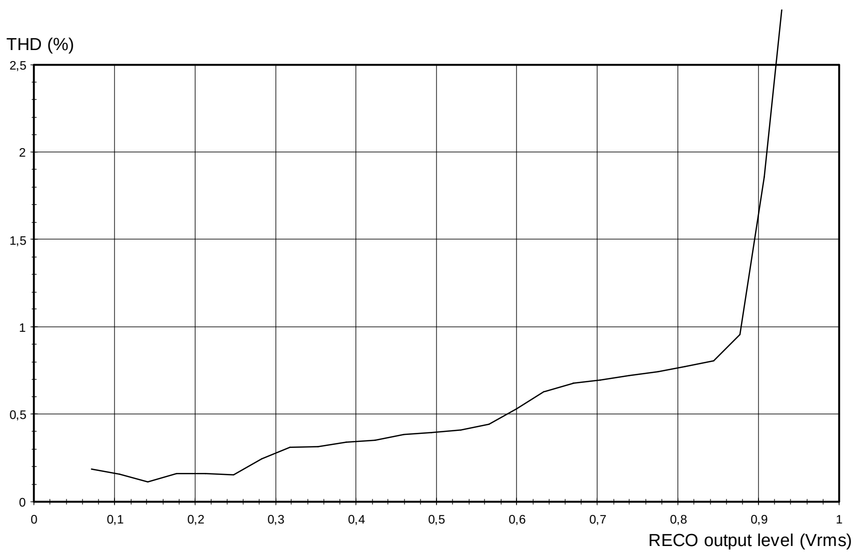

Fig. 37 Distortion on RECO versus level with 5 kΩ load

Fig. 38 shows the noise on RECO loaded with 5 kΩ (psophometrically weighted: P53 curve) as a function of the line current. This curve has been done with selection of the input IR which is left open. With the antisidetone network connected to the input IR, part of the microphone noise generated on the line will be added, but thanks to the low microphone noise value, the effect is almost negligible.

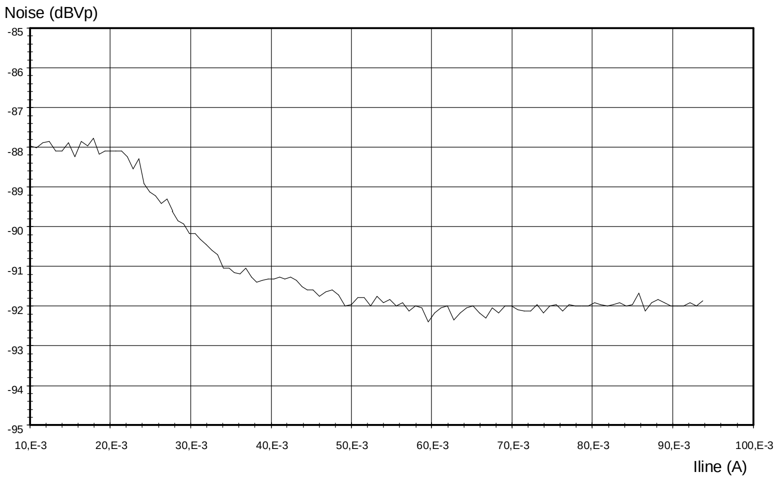

Fig. 38 Noise on RECO

3.4.2 Earphone amplifier QR

Principle of operation

The earphone amplifier of the TEA1099 is able to drive loads down to an impedance of 150 Ω. As can be seen from fig. 32, the earphone amplifier is an amplifier with the gain externally adjustable with a bridge between RECO, GARX and QR. The output is a rail to rail structure suitable for several kind of earpieces and can drive either dynamic, magnetic or piezo-electric earpieces. In case of magnetic or dynamic earpieces, a capacitor in series is required for decoupling; in case of pure capacitive load, a resistor in series is required for stability.

Adjustments and performances

It is possible with a capacitor in series between RECO and GARX to built a high-pass filter and with the capacitor Cgar in parallel with Rgarx to built a low-pass filter. To ensure stability, a capacitor Cgars (Cgars = 10 × Cgar) between pins GARX and GND is necessary. The output is biased at 2Vd with a temperature drift of -4 mV/°C, so the value of VBB affects hardly the output swing capability. When the output is not enabled, there is still an AC path through the gain resistors, its attenuation depends on the value of these resistors (a value of 100 kΩ min is advised for Rgarx). Fig. 39 shows the distortion on QR versus level at Iline = 15 mA on 150 Ω and 470 Ω loads in line powered conditions.

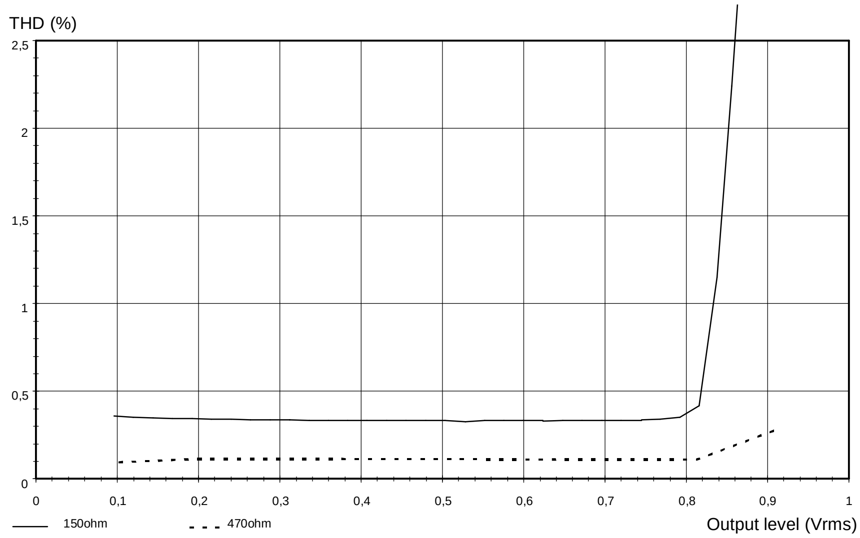

Fig. 39 Distortion on QR versus level

3.4.3 Loudspeaker amplifier LSAO

Principle of operation

As can be seen in fig. 32, the input of the receive channel, pin HFRX is a-symetrical and the signal has to be referenced to GND. The input HFRX is connected, via a decoupling capacitor, to the receive output RECO. In ring mode, the melody signal is directly connected to pin HFRX via a decoupling capacitor. The output of the loudspeaker amplifier is pin LSAO.

As can be seen in fig. 32, the channel to the loudspeaker amplifier is built up out of two parts: a preamplifier and an end-amplifier. The gain of the preamplifier is determined by the duplex controller block. The gain of the end-amplifier is determined by the external feedback resistor Rgals.

The overall gain Gvlsao of the loudspeaker amplifier channel from input HFRX to output LSAO is given as:

$$Gvlsao = 20 \times \log Avls$$

$$Avls = 0.35 \times Rgals / Rstab$$

This gain is not affected by the AGC, using these typical values in the equation, we find a gain equal to:

$$Gvlsao = 20 \times \log Avls = 27.8 \text{ dB} \qquad \text{when } Rgals = 255 \text{ k}\Omega$$

With Rstab being the resistor at STAB of 3.65 kΩ.

Via the volume control input VOL, the volume of the receive signal can be adjusted by the external potentiometer connected to pin VOL. By changing the potentiometer resistance, the gain of the preamplifier varies through the duplex controller. Volume control doesn’t affect the transmit gain in Tx-mode.

The loudspeaker amplifier is enabled only when VBB becomes higher than 2.9 V; when it is “on” it can be automatically disabled if VBB falls down below 2.7 V but this should not happen in normal line conditions because of the dynamic limiter (see §3.2.1).

A ringer mode is available where only the channel from HFRX to LSAO is enabled; this mode can be used with a Switch Mode Power Supply converting the ringing signal into a DC supply applied at pin ESI. In this mode, a square wave melody signal has to be applied at pin HFRX with an advised amplitude of 200 mVpp (lower than 500 mVpp). The volume control is not operating in this mode.

Adjustments and performances

The input signal for the loudspeaker channel has to be coupled via a decoupling capacitor. Together with the input impedance of 20 kΩ at HFRX, a first order high-pass filter is introduced which can be used to adjust the receive curve and/or to reduce any low frequency unwanted signal coming from the line.

The input HFRX is biased at 2 Vd and can handle signal up to 580 mVrms with a total harmonic distortion of 2%.

The output LSAO must be connected to the loudspeaker via a decoupling capacitor. The output is biased at VBB/2 referenced to GND. With the resistor Rgals, the gain of the loudspeaker amplifier channel can be adjusted from +10 to +35 dB. The gain equals typically 27.8 dB with resistor Rgals = 255 kΩ. A capacitor Cgals can be connected in parallel with Rgals to provide a low-pass filter which can be used to adjust the loudspeaker amplifier curve. A capacitor Cgals of at least 150 pF is recommanded in ringer mode.

The output drive capability at pin LSAO is typically 300 mApeak.

The noise level at the output LSAO is -79 dBVp at a gain of 28 dB and with the input HFRX left open.

Out of pin VOL a current Ivol, set by Rstab, is flowing which is proportional to the absolute temperature (PTAT). At room temperature this current is around 5 µA. Together with the resistance of the potentiometer, the current Ivol creates a PTAT voltage on pin VOL. This PTAT voltage is processed by the volume control block; as a result, a temperature independent volume reduction of the output receive signal of 3 dB is obtained at approximately every increase of 1900 Ω of the potentiometer resistance.

This means that a linear potentiometer can be used to control the volume logarithmically, thus in dB. With the advised value of 22 kΩ, the maximum gain reduction of the volume control is more than 30 dB. However, this maximum gain reduction is limited by the switching range, see §3.5. When the resistance of the potentiometer is zero, the receive gain is maximum in Rx-mode.

When digital volume control is desired, the switches can be either MOSFETs or analog switches with very low DC and AC resistance. Due to saturation voltage, it is advised not to use bipolar transistors as switches.

In line powered condition, matching between operating current range and impedance of loudspeaker is important (below approximately 27 mA, a 32 Ω loudspeaker provides more power than a 25 Ω one), Fig. 17 shows the output level on LSAO versus line current on 16 Ω and 50 Ω loads in series with 220 µF in handsfree mode and nominal line powered conditions.

When an external power supply is connected at pin ESI, the LSAO can drive loudspeakers with an impedance down to 8 Ω. In an 8 Ω loudspeaker configuration, the output swing can be optimized by shifting slightly the polarization with a 2.2 MΩ resistor connected between pins GALS and VBB.

Fig. 40 shows the output level (sine wave) on different loads in series with 220 µF in ring mode versus input current at ESI.

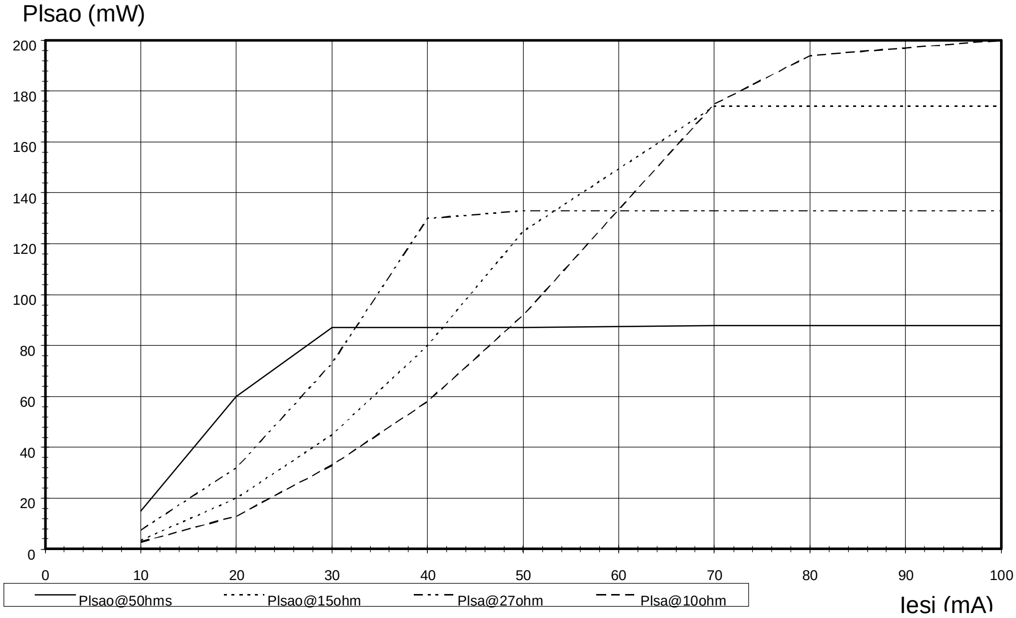

Fig. 40 Output level in ring mode versus input current at ESI

3.4.4 Auxiliary receive amplifier AUXO

Principle of operation

According to the logic selection (see fig. 4), the auxiliary receive amplifier can get signal from four different inputs: IR for line signals, HFRX through the duplex controller in receive, HFTX from TXOUT and the duplex controller (e.g. intercom or answering machine applications) and MIC+/MIC- (e.g. intercom or answering machine applications).

As can be seen from fig. 33, the auxiliary receive amplifier itself is built up out of five parts: four preamplifiers (inputs IR, HFRX,HFTX and MIC+/MIC-) which realize a voltage to current conversion and an end-amplifier which realizes the current to voltage conversion.

The AUXO output of the TEA1099 has an internal impedance of 125 Ω and is able to drive loads down to an impedance of 5 kΩ.

IR has an a-symmetrical high input impedance between pins IR and LN shared with the IR to RECO amplifier. It is equal to 20 kΩ with a maximum tolerance of +/-15%. The overall gain Gv(ir-auxo) of the auxiliary receive amplifier from input IR to output AUXO is given by the equation:

$$Gv(ir\text{-}auxo) = 20 \times \log Avrax$$

$$Avrax = 6.66 \times Rgara/Rrefint \times \alpha$$

with:

Rgara = internal resistor realizing the current to voltage conversion (typically 96.3 kΩ with a spread of +/-15%)

Rrefint = internal resistor determining the current of an internal current stabilizer (typically 14.7 kΩ with a spread of +/- 15% correlated to the spread of Rgara)

α = gain control factor varying from 1 at Iline = 15 mA to 0.49 at Iline = 70 mA when AGC function is applied (see chapter 3.1.4 for details)

Using these typical values in the equation, we find a gain equal to:

Gv(ir-auxo) = 20 $\times$ log Avrax = 32.8 dB at Iline = 15 mA

HFRX has an a-symmetrical high input impedance between pins HFRX and GND shared with the HFRX to LSAO amplifier. It is equal to 20 kΩ with a maximum tolerance of +/-15% and the input is biased at 2 Vd. The gain of the preamplifier is determined by the duplex controller block. The overall max gain Gv(hfrx-auxo) of the auxiliary amplifier from input HFRX to output AUXO is given by the equation:

Gv(hfrx-auxo) = 20 $\times$ log Avreax

Avreax = 0.233 $\times$ Rgara/Rrefint

This gain is not affected by the AGC, using these typical values in the equation, we find a gain equal to:

Gv(hfrx-auxo) = 20 $\times$ log Avreax = +3.7 dB

HFTX has an a-symmetrical high input impedance between pins HFTX and GND shared with the HFTX to LN amplifier. It is equal to 20 kΩ with a maximum tolerance of +/-15% and the input is biased at 2 Vd. The overall gain Gv(hftx-auxo) of the auxiliary amplifier from input HFTX to output AUXO is given by the equation:

Gv(hftx-auxo) = 20 $\times$ log Avbmax

Avbmax = 0.88 $\times$ Rgara/Rrefint

This gain is not affected by the AGC, using these typical values in the equation, we find a gain equal to:

Gv(hftx-auxo) = 20 $\times$ log Avbmax = 15.2 dB

MIC+/MIC- has symmetrical high input impedances (typically 70 kΩ -2 times 35 kΩ- between pins MIC+ and MIC- with maximum tolerances of +/- 15%) shared with the MIC+/MIC- to LN amplifier. It is equal to 20 kΩ with a maximum tolerance of +/-15%. The overall gain Gv(mic-auxo) of the auxiliary handset microphone amplifier from input MIC+/MIC- to output AUXO is given by the equation:

Gv(mic-auxo) = 20 $\times$ log Avhmax

Avhmax = 2.78 $\times$ Rgara/Rrefint

This gain is not affected by the AGC, using these typical values in the equation, we find a gain equal to:

Gv(mic-auxo) = 20 $\times$ log Avhmax = 25.2dB

Adjustments and performances

32.8 dB of receive gain between IR and AUXO compensate approximately the attenuation provided by the antisidetone network.

Fig. 41 shows the frequency response of the auxiliary receive amplifier from IR to AUXO at different temperatures.

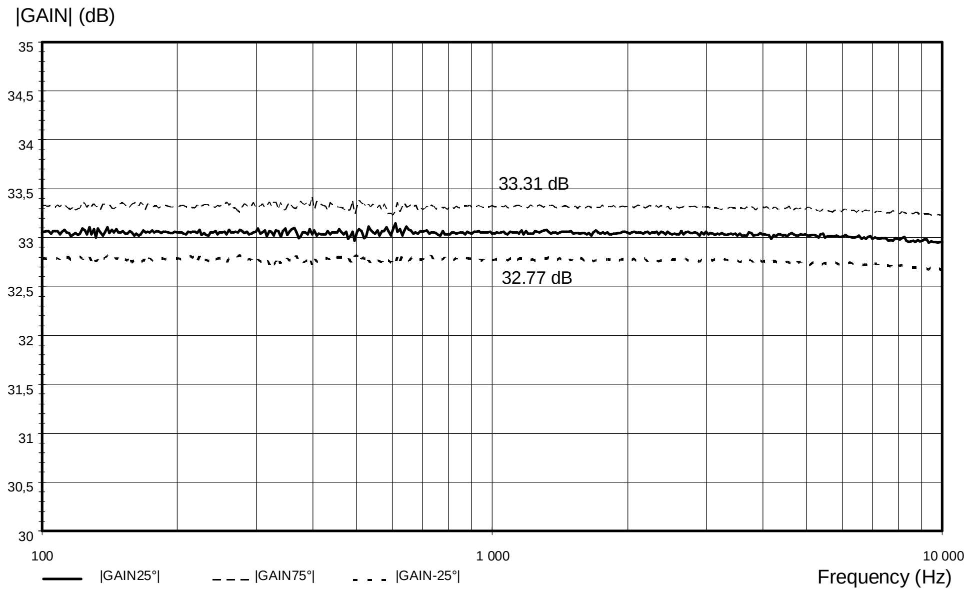

Fig. 41 Auxiliary receive gain versus frequency, influence of temperature

The output is biased at 2 Vd with a temperature drift of -4 mV/°C, so the maximum output swing on AUXO depends hardly on the value of VBB.

The receiving input IR can handle signals up to 50 mVrms with less than 2% THD. Fig. 42 shows the distortion on AUXO for a line current equal to 75 mA.

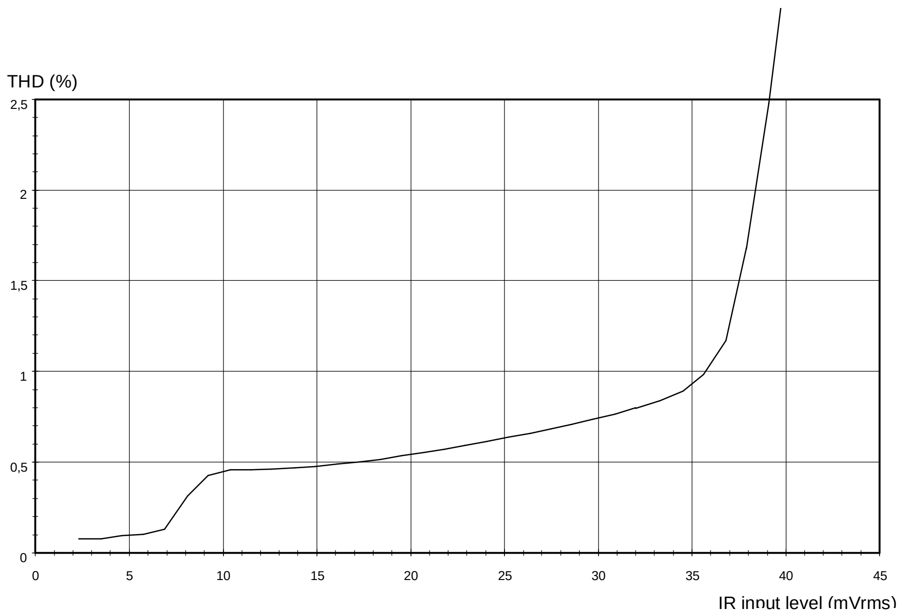

Fig. 42 Distortion on AUXO versus input signal on IR

The input HFRX is biased at 2 Vd and can handle signals up to 580mVrms with less than 2% THD. Fig. 43 shows the distortion on AUXO when the limitation is related to the input voltage at HFRX for a line current equal to 75 mA.

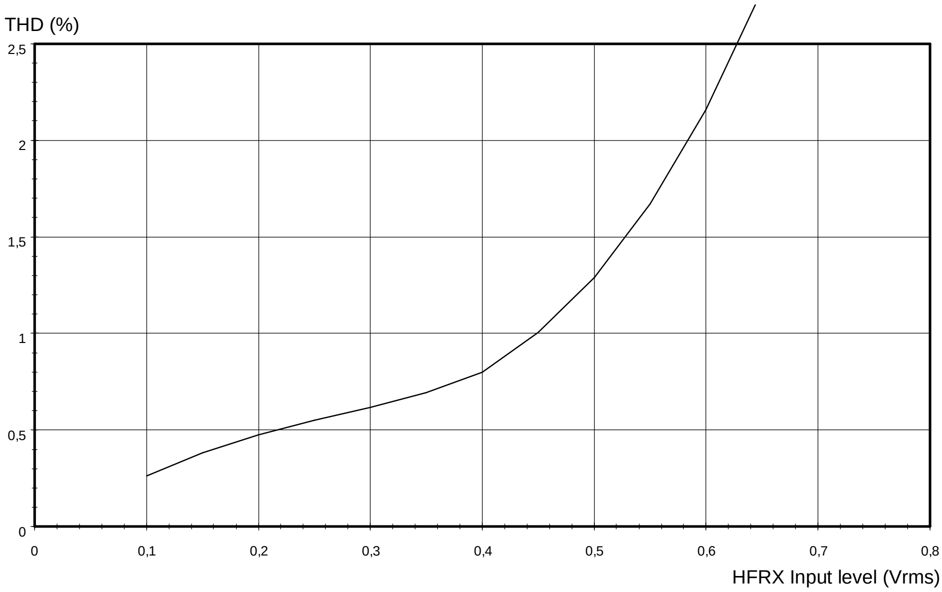

Fig. 43 Distortion on AUXO versus input signal on HFRX

The input HFTX is biased at 2 Vd and can handle signals up to 140 mVrms with less than 2% THD. Fig. 44 shows the distortion on AUXO for a line current equal to 75 mA.

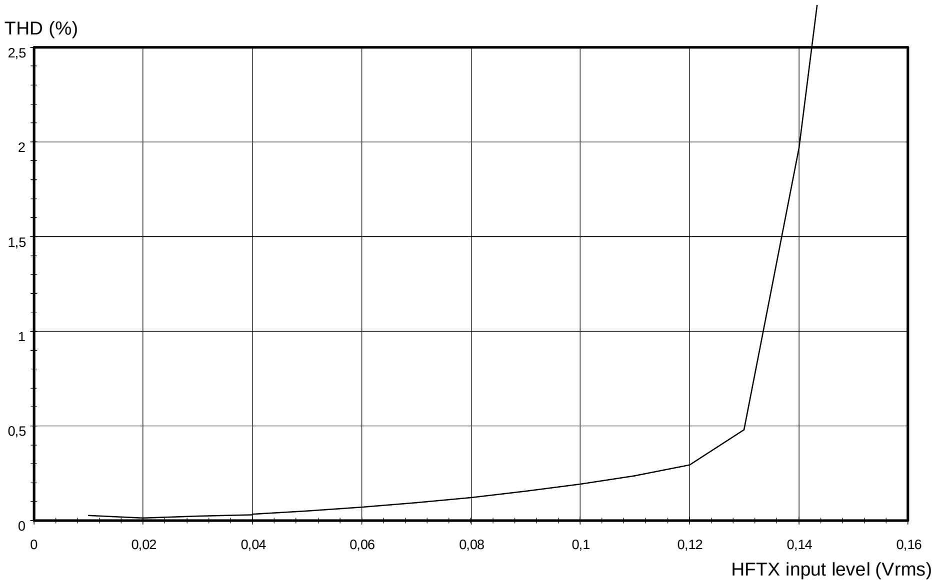

Fig. 44 Distortion on AUXO versus input signal on HFTX

Fig. 45 shows the distortion of the signal on AUXO as a function of the rms signal on AUXO with a load of 5 kΩ and a line current of 15 mA.

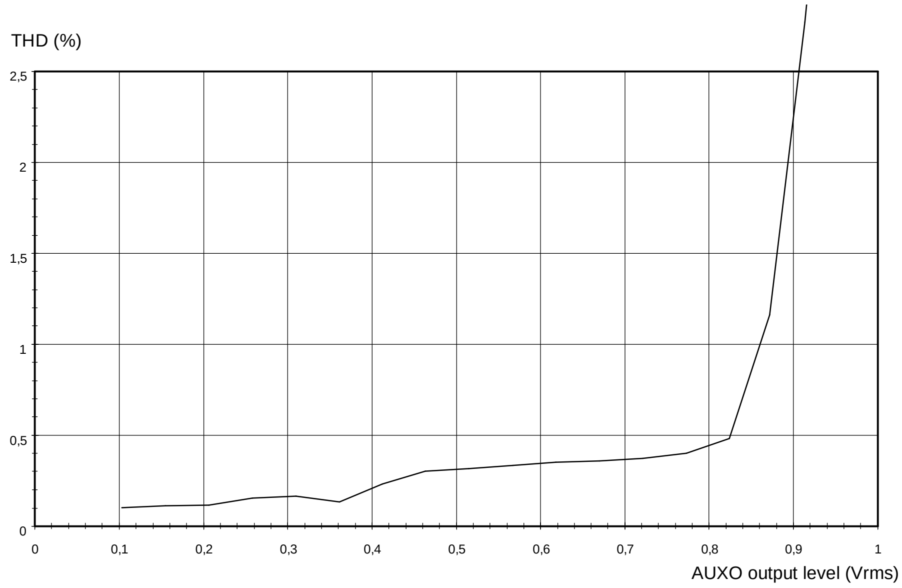

Fig. 45 Distortion on AUXO versus level with 5 kΩ load

Fig. 46 shows the noise on AUXO loaded with 5 kΩ (psophometrically weighted: P53 curve) as a function of the line current. This curve has been done with selection of the input IR which is left open. With the antisidetone network connected to the input IR, part of the microphone noise generated on the line will be added but, thanks to the low microphone noise value, the effect is almost negligible. Fig. 47 shows the noise on AUXO when the input is from MIC+/MIC-.

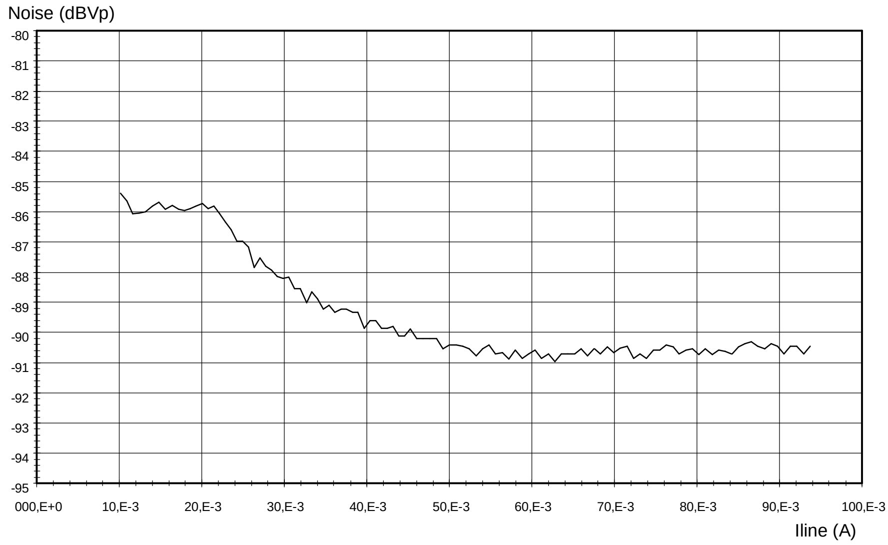

Fig. 46 Noise on AUXO with input at IR

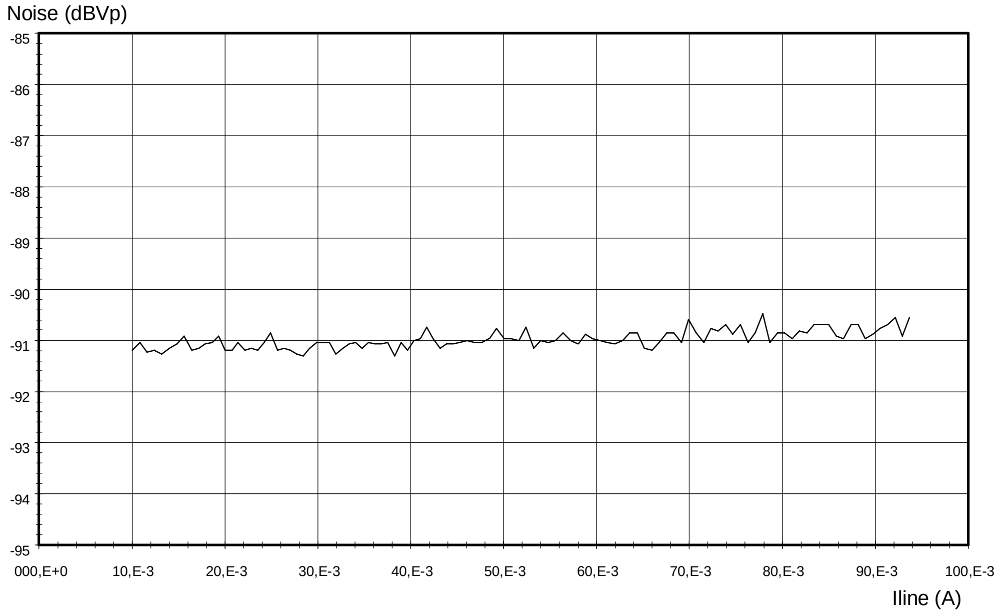

Fig. 47 Noise on AUXO with input at MIC+/MIC-

3.4.5 Auxiliary microphone monitor amplifier

When Group-listening mode is selected, the microphone signal is monitored with a fixed gain of 49.8 dB and a spread of +/- 2.5 dB at the output TXOUT. In this condition, the end-amplifier of the handsfree microphone channel is switched into high impedance output mode. The channel between MIC+/MIC- and TXOUT is built up out of two parts: a preamplifier which makes a voltage to current conversion followed by an end-amplifier which realizes the current to voltage conversion.

The output is biased at 2 Vd and can drive up to +/- 300 µA rms and the maximum output swing is VBB-0.8V.

An external antihowling circuit is shown in §5 fig. 63 while the block diagram of the Group-listenning application is shown fig. 57.

3.5 Duplex controller

In this chapter, the principle of operation of the duplex controller will be described as well as its adjustments and performances. This will be done with the help of fig. 48.

Nevertheless,the duplex controller part is enabled only when VBB becomes higher than 2.9 V (see §3.2.1).

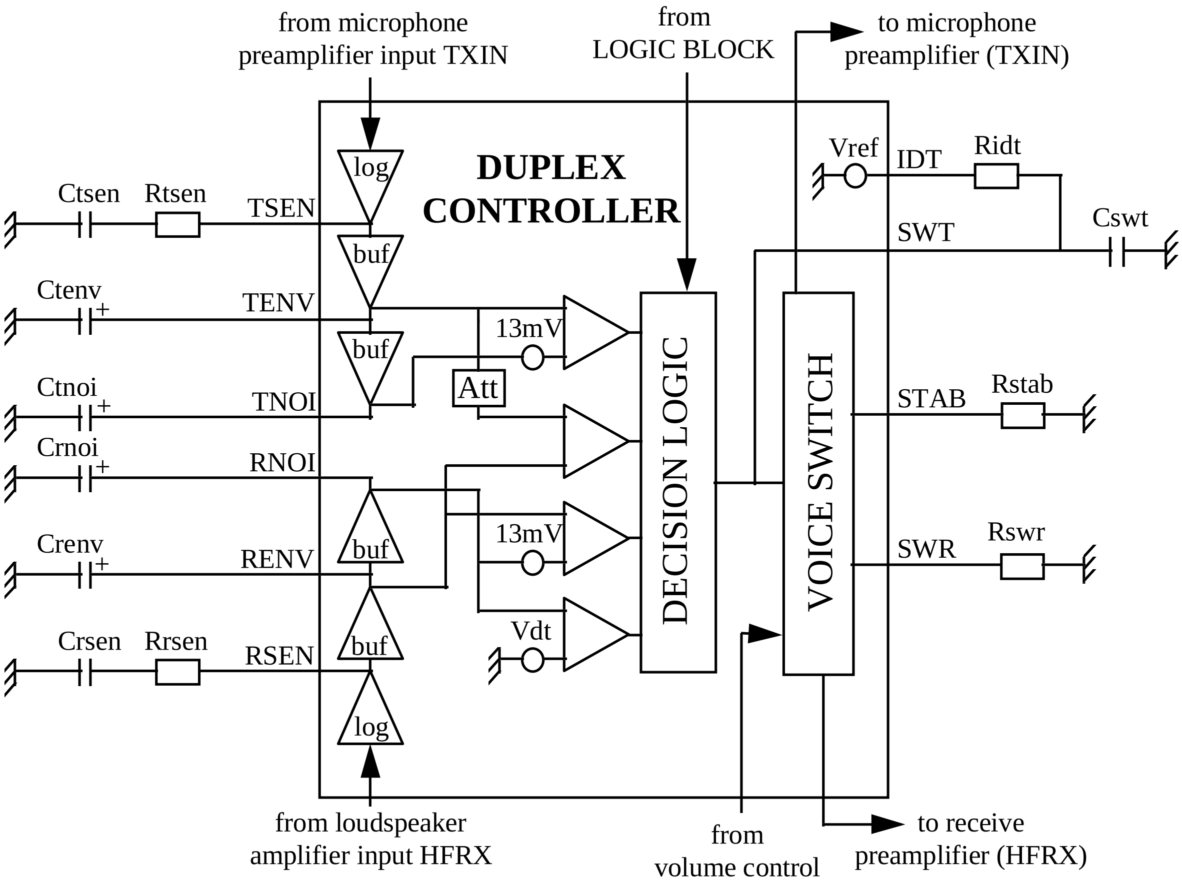

Fig. 48 Principle of the duplex controller

As can be seen in fig. 48, the duplex controller is built up out of signal and noise envelope detectors, decision logic and a voice switch.

The signal and noise envelope detectors determine the signal envelopes and the noise envelopes of both the transmit and receive signal. These envelopes are used by the decision logic to determine into which mode the TEA1099 has to switch over (Tx, Rx or Ix-mode). The logic charges and discharges the capacitor Cswt and the resulting voltage on pin SWT controls the voice switch. The voice switch switches over the TEA1099 between the three modes while keeping the loop gain constant.

In paragraphs 3.5.1 to 3.5.3, the principle of operation of the three parts is given. In paragraph 3.5.4, the adjustments and performances of the complete duplex controller are given.

3.5.1 Signal and noise envelope detectors

The signal and noise monitors of the transmit and receive channels are globally the same.

Therefore, the principle of the detectors will be explained with the help of one of them: the signal and noise detector of the transmit channel.

The microphone signal on pin TXIN is sent to the first stage of the detector, see fig. 48. The first stage amplifies the microphone signal from pin TXIN to pin TSEN with an internal gain of 40 dB. Via the RC combination RtsenCtsen, the signal on TSEN is converted into a current. This conversion determines the sensitivity of the envelope detector. The current is logarithmically compressed and internally converted to a voltage which represents the compressed microphone signal. At room temperature, an increase of the microphone signal with a factor of 2 will increase the signal envelope with 18 mV if the current through TSEN stays between 0.8 and 160 µArms. Outside this region the compression is less accurate.

The compressed microphone signal is buffered by the second stage to pin TENV. As the buffer can source 120µA and sink 1µA, the signal on TENV follows the positive peaks of the compressed signal, this signal is called the signal envelope. The time constants of the signal envelope are therefore determined by the combination of the internal current sources and the capacitor Ctenv.

The voltage on TENV is buffered by the third stage to pin TNOI. As this buffer can source 1 µA and sink 120 µA, the signal on TNOI follows the negative peaks of the signal on TENV. This is called the noise envelope because it represents the background noise. The time constants of the noise envelope are determined by the combination of the internal current sources and the capacitor Ctnoi. Both capacitors Ctnoi and Crnoi are provided with a start-up circuit. During start-up the capacitors are charged with approximately 40 µA up to 1.9 V. The starter will restart when the voltage on the capacitors drops below 0.9 V.

As can be seen in fig. 48, the principle of operation of the signal and noise envelope detectors of the receive channel is equal to the one of the transmit channel. However, the gain of the first stage (input to pin RSEN) is 0 dB instead of 40 dB for the transmit channel, this is in order to compensate the level on TXIN which is not yet amplified.

The behavior of the envelopes is illustrated in fig. 49 where the signal and noise envelope of one channel are depicted together with the input signal.

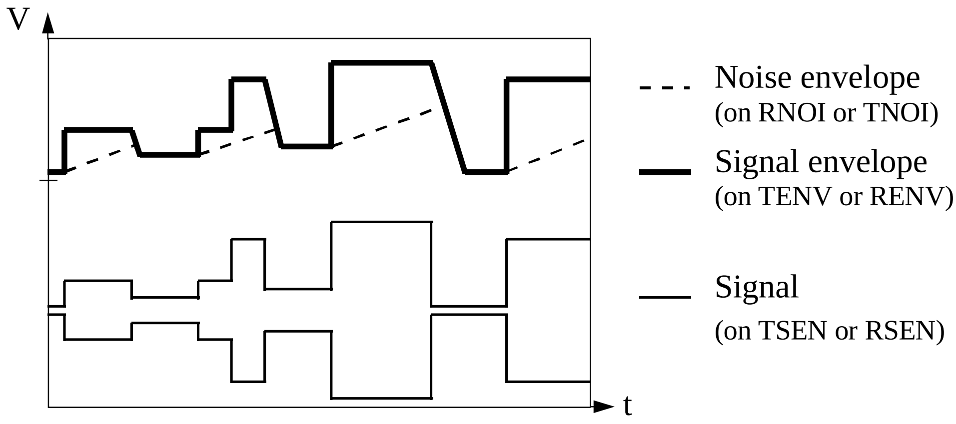

Fig. 49 Typical behavior of the signal and noise detectors

In fig. 49 it is shown that when the input signal raises quickly, the envelope signal follows immediately and the noise envelope slowly follows the envelope signal. When the input signal decreases, the envelope signal follows immediatly but nevertheless less quickly than when it raises, the noise envelope follows immediatly the decrease of the envelope signal and never crosses it.

3.5.2 Decision logic

The signal and noise envelopes of the transmit and receive signal are used by the decision logic to determine in which mode the TEA1099 has to be.

The output of the logic is a current source which charges or discharges the capacitor Cswt at pin SWT. If the logic determines Tx-mode, the capacitor Cswt is discharged with 10 µA. When Rx-mode is determined, Cswt is charged with 10 µA. When Ix-mode is determined, the current source is zero and the voltage on SWT becomes equal to the voltage on pin IDT via the current provided through the resistor Ridt. The time constants of the duplex controller are therefore determined by the combination of the internal current sources, the capacitor Cswt and the resistor Ridt.

As can be seen in fig. 48, the envelopes are not used directly by the decision logic.

First, to have a clear choice between signal and noise, the signal is considered as speech when its envelope is more than 4.3 dB above the noise envelope. At room temperature, this is equal to a voltage difference of 13 mV. This so called speech/noise threshold is implemented in both the receive and the transmit channel. At the end of paragraph 3.5 (miscellaneous) a way to increase this threshold is discussed.

Second, the signal on TXIN contains the signal of the local talker as well as the signal coming from the loudspeaker (acoustic coupling). In Rx-mode, the contribution of the loudspeaker overrules the contribution of the local talker. As a result, the signal envelope on TENV is mainly formed by the loudspeaker signal, to correct this, an attenuator is placed between TENV and the TENV/RENV comparator. The attenuation equals the attenuation applied to the microphone amplifier gain. Thus when the TEA1099 is in Rx-mode, the attenuation equals the switching range.

Third, when a dial tone is present on the line, without measures this would be recognized as noise after some delay because its level is constant. As a result, the TEA1099 would go to Ix-mode and the user of the set would hear the dial tone fade away. Therefore, a dial tone detector is incorporated which doesn't consider input signals as noise when they have a level higher than the dial tone level. The dial tone level, represented by Vdt in fig. 48, is adjustable by Rrsen.

In a similar way, as the coupling between the loudspeaker and the microphone is very strong a too high level of noise would be monitored at pin TNOI when dial tone is received. This is prevented by clamping internally the level of noise that can be monitored on TNOI to a realistic value (0.75 mVrms at TXIN with Rtsen = 10 kΩ).

When these three corrections are made, the signal and noise envelopes are used by the comparators and the logic. As already explained, the output of the logic is a current source. The relation between the current source and the output of the comparators is given in the table of fig. 50. If for instance, TENV>RENV (transmit signal larger than receive signal) and TENV>TNOI (transmit signal more than 4.3 dB larger than noise level), then the output current will be -10 µA.

Comparator TENV/TNOI	1	x	x	0	x
Comparator TENV/RENV	1	0	0	1	0
Comparator RENV/RNOI	x	1	x	x	0
Comparator RNOI/Vdt	x	x	1	x	0
Output current	-10µA	+10µA	+10µA	0µA	0µA

Fig. 50 Truth table of the decision logic

When pin DLC is forced to GND the output current is forced to be -10 µA, which forces the TEA1099 into Tx-mode and mutes the receive path. The voltage on pin SWT is internally limited to IDT-0.4 V and IDT+0.4 V.

3.5.3 Voice switch

With the voltage on pin SWT, the voice switch regulates the gain of the microphone preamplifier and the receive channel preamplifier in such a way that the sum of the transmit and receive gain is kept constant. This is done to keep the loop gain of the handsfree telephone set constant, see also the introduction §1. The switch-over behavior of the voice switch is shown in fig. 51.

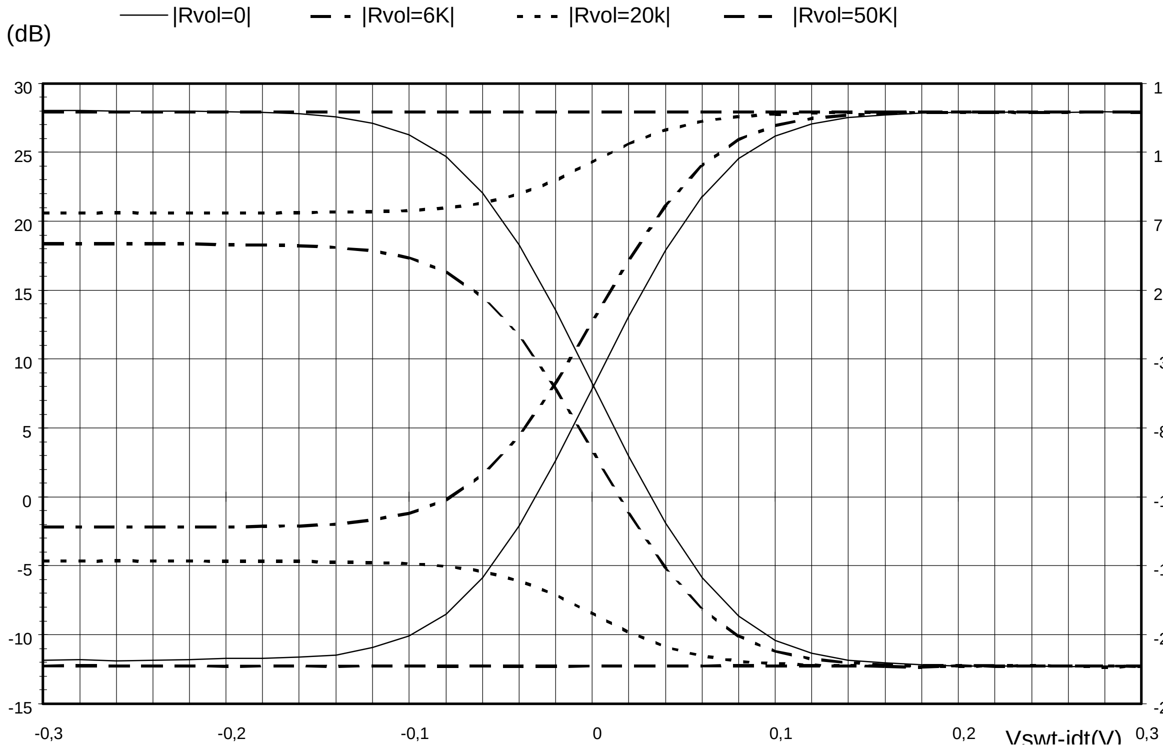

Fig. 51 Behavior of the voice switch

When the voltage on SWT is more than 180 mV below the voltage on IDT, the TEA1099 is fully switched to Tx-mode (gain of the transmit path at maximum and gain of the receive path at minimum). When the voltage on SWT is more than 180 mV above the voltage on IDT, the TEA1099 is fully switched to Rx-mode (gain of the receive path at maximum and gain of the transmit path at minimum). The TEA1099 is considered to be in Ix-mode when the voltage on SWT equals the voltage on IDT. When the capacitor Cswt is charged or discharged, the voltage on SWT varies and as a result the voice switch will smoothly switch over between the modes keeping the sum of the transmit and receive gains constant.

The difference between the maximum and the minimum gain of the receive or transmit preamplifiers is called the switching range. This range is determined by the ratio of Rswr and Rstab, see paragraph 3.5.4. Both Rswr and Rstab set internally used reference currents which are proportional to absolute temperature (PTAT).

As already stated in §3.4.3 the volume control acts upon the receive preamplifier via the control of the voice switch. As a result, the loop gain of the handsfree set is kept constant when the volume of the receive path is adjusted. However, the voice switch is designed such that the volume control has no influence in Tx-mode. In the extreme case, when the volume of the receive channel is reduced with the value of the switching range, the TEA1099 virtually does not switch over. In order to avoid inversion of the gain in Rx-mode, the volume control range of the TEA1099 cannot be larger than the switching range.

3.5.4 Adjustments and performances of the duplex controller

The adjustment of the duplex controller has to be performed according to the following receipe:

1. Determine the switching range
2. Determine dial tone detector level

3. Determine sensitivity
4. Determine timings

Ad 1. Determine switching range

The switching range Asw is determined by the ratio of the two resistors Rswr and Rstab according to:

$$Asw\ (dB) = 20 * \log\ (Rswr / Rstab)$$

The resistor Rstab has to be 3.65 kΩ. The value of the resistor Rswr can vary reasonably between 20 kΩ and 1.5 MΩ resulting in a switching range between 15 dB and 52 dB. With Rswr of 365 kΩ, the switching range is typically set to 40 dB.

The switching range is calculated out of the loop gain (Aloop). In a handsfree application, the loop gain has to be smaller than one (<0 dB) and can be calculated as follows:

Aloop = Atx1099 + Ast + Arx1099 + Aac - Asw.

with Atx1099 = sending gain of the TEA1099 (TXIN to TXOUT + HFTX to LN)

Ast = electrical sidetone

Arx1099 = receive gain of the TEA1099 (LN to IR to LSAO)

Aac = electro-acoustic coupling from loudspeaker to microphone (LSAO to TXIN)

Asw = switching range

In this calculation, the worst case has to be taken for Ast and Aac. Furthermore, for safety, it is advised to choose Asw large enough to compensate spreads (margin from 8 to 15 dB).

The electrical sidetone is the difference (in dB) between the wanted receive signal on the TEA1099 and the unwanted part of the transmit signal received while having an equal signal level on pin LN for both the transmit and the receive signal. Ast is dependent of frequency and connecting conditions of the set (line length, line impedance).

The acoustic coupling is dependent on the environment of the telephone set, for the determination of Aac, the worst condition has to be searched.

If a certain minimum volume control range is required, the switching range must not be chosen smaller.

It is also possible to determine the switching range by experiments:

As for the calculation, it is necessary to identify what the worst conditions are for sidetone and acoustic coupling. In these worst conditions, Rswr can be adjusted in such a way that the handsfree telephone set is at the limit of howling. Then the determined value of Rswr must be increased in order to have a margin of 8 dB to 15 dB.

Handsfree behavior will be more comfortable for the user if the switching range is not too large. So, **it is advised to take care of the acoustic coupling between the loudspeaker and the microphone** which might come from the cabinet of the terminal itself.

Ad 2. Determine dial tone detector level

The dial tone detector level on HFRX is determined by the value of Rrsen according to:

Vdialtone = 2.5 µA * Rrsen

With Rrsen of 10 kΩ, the dial tone detector level will be 25 mVrms. This means, a continuous signal on the input HFRX larger than 25 mVrms will be recognized as a dial tone.

Ad 3. Determine sensitivity

The sensitivity is set by Rrsen and Rtsen. The resistor Rrsen is already determined by the dial tone detector level. It must however be checked if the chosen value for Rrsen is a practical one for the dynamic range of the logarithmic compressor. The optimized range for the compression is when the current flowing through pin RSEN is between 0.8 to 160 µArms. This means that at nominal receiving signal the current through RSEN is preferably around 11 µArms. This gives a maximum dynamic range of plus and minus 23 dB.

The same counts for pin TSEN.

The resistor Rtsen has to be preferably chosen in such a way that both channels have the same priority for the duplex controller. This can be obtained by choosing Rtsen according to:

20 * log (Rtsen) = 20 * log (Rrsen) - Atx1099 - Ast - Arx1099 + Atsen + 1/2 Aloop

with Atsen = internal gain from TXIN to TSEN = 40 dB.

In this formula, the maximum loop gain and the worst case sidetone are used. If it is preferred to give the transmit channel priority above the receive channel, the value of Rtsen has to be chosen smaller. For the opposite, the value of Rtsen has to be chosen larger. With respect to the calculated setting, Rtsen and Rrsen can be varied with plus and minus 1/2 * Aloop (in dB).

The capacitors Ctsen and Crsen form first order high-pass filters respectively with Rtsen and Rrsen to reduce influence of low frequencies on the switching behavior. It is suggested to choose the capacitors Ctsen and Crsen such that the cut-off frequencies of the filters are similar.

When the calculated sensitivity setting is implemented, subjective tests with real telephone lines will be necessary to come to the optimal sensitivity setting.

Once Rrsen is determined, it would also be possible to determine Rtsen only by experiments. In this case, subjective tests with different line conditions (attenuation, impedance, length) have to be carried-out until the optimal sensitivity setting is found.

Ad 4. Determine timings

The timings which can be set are : signal envelope timing and noise envelope timing for both channels, switch-over timing and idling timing.

The signal envelope timing is set by the capacitors Ctenv and Crenv. Because of the logarithmic compression between TSEN and TENV respectively RSEN and RENV, the timing can be expressed in dB/ms. At room temperature, the following relation counts :

Timing ≅ I / (3 * C)   (in dB/ms)

With I = charge or discharge current from pin TENV, RENV, TNOI, RNOI

C = timing capacitor Ctenv, Crenv, Ctnoi, Crnoi

With the advisable signal envelope timing capacitors Ctenv and Crenv of 470 nF, the maximum attack-timing of the signal envelopes will be around 85 dB/ms (I=120 µA). This is enough to track normal speech. The release timing will be 0.7 dB/ms (I=1 µA). This is enough to smoothen the signal envelope and to eliminate the influence of room echoes on the switching behavior.

With the advisable noise envelope timing capacitors Ctnoi and Crnoi of 4.7 µF, the attack timing of the noise envelopes will be 0.07 dB/ms (I=1 µA). This is small enough to track background noise and not to be influenced by speech bursts. The maximum release timing will be 8.5 dB/ms (I=120 µA). This is enough to track the signal envelope during release because the signal envelope release timing is 0.7 dB/ms which is a factor smaller. It is advised to choose the signal envelope timing and the noise envelope timing of both channels equal for optimum operation of the duplex controller.

The switch-over timing is determined by the value of the switch-over capacitor Cswt. The idling timing is determined by the combination of Cswt and the idling resistor Ridt.

The output current of pin SWT is Iswt, a voltage difference over Cswt can be obtained according to :

$$\delta Vswt/t = Iswt / Cswt \text{ (mV/ms)}.$$

With the advised value of 220 nF for Cswt, the obtained voltage difference is 45 mV/ms. The switch-over time is dependent on the voltage difference which has to be generated on pin SWT. Assuming the set is in full Tx-mode, then the voltage on SWT will be V(IDT)-400 mV, see fig. 51. To reach Rx-mode a voltage difference of 580 mV must be generated to end up a voltage of V(IDT)+180 mV. So in this case the switch-over time will be 13 ms. When the set is in Ix-mode, the voltage on SWT equals the voltage on IDT, in that case switching to Tx-mode or to Rx-mode requires a voltage generation of only 180 mV and they will be reached in 4ms.

The idling timing is determined by an RC time constant. It is supposed that Ix-mode is reached when a time (tidt) is elapsed:

$$tidt = 4 * Ridt * Cswt$$

With the advised value for Ridt of 2.2 MΩ, an idling time of around 2 seconds is obtained. To have a clearly determined idling timing, it is advised not to use a capacitor with a high leakage current.

When the calculated timing settings are implemented, subjective tests with real telephone lines will be necessary to be sure that the optimal timings have been set.

Miscellaneous

Idle-mode gain

When a handsfree telephone set is used at one end of the line and a conventional set at the other end, the user of the conventional set may think that the line is cut when the handsfree set stays in receive mode while no signal on the line is present. This is avoided when the handsfree set switches over to idle mode. This mode is incorporated in the TEA1099 and is placed exactly at mid attenuation between transmit and receive mode. When it is desired to have an idle mode which is closer to transmit than receive mode, the circuit of fig. 52 can be applied.

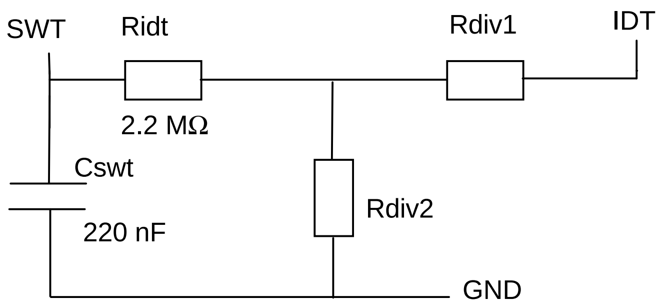

Fig. 52 Circuit for shifting the idle mode

With the circuit of fig. 52, in idle mode, the voltage on SWT will not go to the voltage on IDT but to the voltage on IDT minus the voltage drop over Rdiv1. The voltage drop over Rdiv1 determines the shift of the idle mode (in dB). This shift can be read from fig. 51 , when the voltage drop over Rdiv1 is taken as the X-axis value. The voltage on IDT is approximately 1.2 V, so with for instance Rdiv1 = 33 kΩ and Rdiv2 = 1 MΩ, the shift will be about 10 dB. It is advised not to choose Rdiv2 lower than 1Mohms in order to limit the current drawn from IDT. By connecting Rdiv2 to VBB instead of GND, the idle mode is shifted towards the receive mode.

Signal to Noise ratio

In noisy environments, like offices, a handsfree set can show an unsteady behavior in idle mode (unwanted switching over from Ix to Tx-mode). In the TEA1099, this unsteady behavior is reduced by the implemented speech/noise threshold of 4.3 dB. However, when a larger threshold is required, this can be achieved by connecting a resistor Rtnoi in series with Ctnoi.

When there is only noise present at the input of the envelope detector, the voltages on pins TENV and TNOI are equal. When suddenly, a signal is present, the level on TENV will increase. Without Rtnoi, the voltage on TNOI will increase only slowly because of the charging current of 1 µA. When a resistor Rtnoi is placed in series with Ctnoi, under the same conditions, this 1 µA current will cause a voltage jump on TNOI. This jump determines the shift of the speech /noise threshold. As explained in §3.5.2, at room temperature, the 4.3 dB threshold equals 13 mV. A resistor Rtnoi in series with Ctnoi will add an extra voltage to this threshold of 1 µA × Rtnoi. When for instance, a resistor of 10 kΩ is chosen for Rtnoi, the speech/noise is increased to 23 mV which is equal to 7.6 dB at room temperature. The new speech/noise threshold is slightly dependent on temperature and on the spread of the internal current source and therefore less accurate than the internal 4.3 dB. It is advised not to use a resistor larger than 12 kΩ.

Ground layout

The layout of the ground is very important for noise and for the duplex controller behavior. In fact, high currents generate residual voltages on the PCB and these voltages may affect dramatically some references or may couple loudspeaker signal with microphone. Fig. 53 gives a reference for the connection of different components to the grounds.

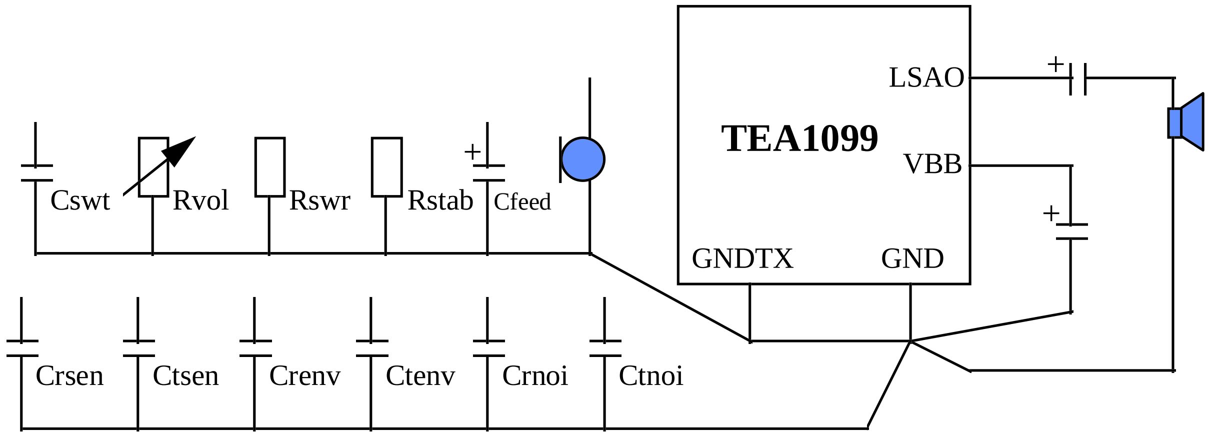

Fig. 53 GND and GNDTX connections

Antihowling

In group-listening application there is an acoustic coupling between the loudspeaker and the handset microphone. When the microphone is too close to the loudspeaker, the gain of the loop becomes higher than 1 and howling occurs. This howling may disturb the other party, specially if it is an operator with a headset. Around the TEA1099 it is possible to build an antihowling circuit which limits this howling. For this purpose, in group-listening mode, the microphone signal is amplified by 49.8 dB at pin TXOUT and the gain of the loudspeaker amplifier can be reduced by pulling pin DLC to GND. The solution advised for this antihowling circuit is to measure the amplitude of the signal at pin TXOUT, when it is too high for a too long time pull the pin DLC to GND. This can be done with a very simple circuit where the time constants are set by capacitors (see fig. 63).

Ringer mode

In the ringer mode provided, a switch mode power supply circuit (DC/DC converter) has to be added to convert the input ringing signal into a DC supply connected at pin ESI. The current coming from this supply is usually varying with the frequency of the input signal thus modulating the melody with this low frequency. It is possible to reduce the acoustic effect of this modulation by slowing down the dynamic limiter. This can be done by increasing the value of Cdlc in ringer mode from 0.47 µF to 10 µF (switching a parallel 10 µF capacitor by means of a DMOS transistor).

3.6 Logic block

3.6.1 Logic inputs

In this chapter the 5 logic inputs which control the TEA1099 are described. The selected channels for each combination of the logic inputs are depicted in fig. 54. When a channel is selected, the relevant preamplifier is switched-on and all the others connected to the same end-amplifier are switched-off. The end-amplifiers, the duplex controller block and the electret microphone supply are also enabled or disabled according to fig.54.

In order to guarantee handset conversation mode when an other set is connected in parallel and when the line current is so low that the dialler or the microcontroller can't operate, the values of the logic inputs are ignored when VBB has not been higher than 2.9 V or has fallen down below 2.5 V and the handset conversation mode is forced (see §3.2.1).

Input PD/

This input is active low, it can be driven by an open drain structure because a pull-up to VBB is included. Nevertheless, in case of I/O structure on the microcontroller side, a push-pull output structure is recommended to polarize properly the input of the microcontroller when VBB varies (no current will flow from VDD to VBB via this pin). The threshold voltage level is 0.65 V typically with a temperature coefficient of -2 mV/°C. The input voltage must stay within the limits GND -0.4 V to VBB +0.4 V.

Except in ring mode (see fig. 54), when PD/ is low all the internal consumptions are switched-off, only the supply block VDD is kept active in order to sink current from the tank capacitor connected at VBB.

Input MUTT/

This input is active low, it can be driven by an open drain structure because a pull-up to VBB is included. Nevertheless, in case of I/O structure on the microcontroller side, a push-pull output structure is recommended to polarize properly the input of the microcontroller when VBB varies (no current will flow from VDD to VBB via this pin). The threshold voltage level is 0.65 V typically with a temperature coefficient of -2 mV/°C. The input voltage must stay within the limits GND -0.4 V to VBB +0.4 V.

Input MUTR/

This input is active low, it must be driven by a push-pull structure, the threshold voltage level is 0.65 V typically with a temperature coefficient of -2 mV/°C. The input voltage must stay within the limits GND -0.4 V to VBB +0.4 V.

When using a standard dialler for only basic applications: handset mode, handsfree mode and dialling modes, this input has to be connected to MUTT/ input. In this configuration, when MUTT/, MUTR/ are low the TEA1099 is switched to DTMF dialling mode.

Input HFC

This input is active high, it can be driven by an open drain structure because the pull-down is included, the threshold voltage level is 1.3 V typically with a temperature coefficient of -4 mV/°C. The input voltage must stay within the limits GND -0.4 V to VBB +0.4 V.

When active the input HFC switches-on the duplex controller block if the supply condition enables it (see §3.2.1).

If the supply conditions are such that the handset conversation mode is forced, this pin becomes an output at logic level “0” with an output sink capability of 300 µA.

Input AUXC

This input is active high, it can be driven by an open drain structure because the pull-down is included, the threshold voltage level is 0.65 V typically with a temperature coefficient of -2 mV/°C. The input voltage must stay within the limits GND -0.4 V to VBB +0.4 V.

When using a standard dialler for only basic applications: handset mode, handsfree mode and dialling modes, this input can be left open.

3.6.2 Connections

The following table gives the details of the connections which are enabled according to the logic inputs and some ideas of applications which are foreseen.

	LOGIC INPUTS					CONNECTIONS	APPLICATIONS
	(PD/)	HFC	MUTT/	MUTR/	AUXC		
1	0	X	X	X	1	HFRX -> LSAO	Ring mode with SMPS
2	0	X	X	X	0		Flash, DC dialling
3	1	0	0	0	0	DTMF->LN ; CT->RECO (MICS, QR)	Tel. Set: DTMF dialling
4	1	0	0	1	0	MIC->AUXO; RAUX->RECO (MICS, QR)	Cordless intercom with handset
5	1	0	0	1	1	MIC->AUXO; RAUX->RECO (QR)	
6	1	0	1	X	0	MIC->LN; IR->RECO; IR->AUXO MIC->TXOUT (MICS, QR)	Handset conversation; A.M. (record conv.)
7	1	0	1	1	1	MIC -> LN; MIC -> TXOUT (QR)	
8	1	0	1	0	1	TXAUX->LN ; IR->AUXO	Conversation with auxiliary (Fax, A.M., RF interface, ..)
9	1	0	0	0	1	TXAUX->LN ; IR->AUXO RAUX -> RECO	
10	1	1	1	1	1	TXIN->TXOUT; HFTX->LN; IR->RECO; HFRX->AUXO	Cordless: Handsfree in mobile
11	1	1	0	1	1	RAUX->RECO; HFRX->LSAO	Listenning on loudspeaker
12	1	1	0	0	1	TXAUX->LN ; IR->AUXO; RAUX->RECO; HFRX->LSAO	Answering Machine (outgoing message)
13	1	1	0	0	0	DTMF->LN ; CT->RECO; HFRX->LSAO (MICS, QR)	HF/GL Tel Set DTMF dialling
14	1	1	1	0	1	TXAUX->LN ; IR->AUXO; IR->RECO; HFRX->LSAO	Answering machine (incoming message); Fax with monitoring
15	1	1	0	1	0	TXIN->TXOUT; HFTX->AUXO; RAUX->RECO; HFRX->LSAO (MICS)	Cordless intercom with base; A.M. (record or listen with base)
16	1	1	1	1	0	TXIN->TXOUT; HFTX->LN; IR->RECO; IR->AUXO; HFRX->LSAO (MICS)	Handsfree conversation; A.M. (record conv.)
17	1	1	1	0	0	MIC->LN; IR->RECO; IR->AUXO; HFRX->LSAO MIC->TXOUT (MICS,QR)	Group-listenning conversation; A.M. (record conv.)

Fig. 54 Table of connections

In modes 1 and 10, the volume control doesn't operate.

In modes 11, 12, 13, 14 and 17, the microphone channel of the duplex controller is muted and the receive mode is forced.

In mode 15, by connecting pin DLC to ground, it is possible to force the duplex controller in transmit mode.

When VBB has not reached a value higher than 2.9 V or has fallen down below 2. 5 V, the logic inputs are ignored and the handset conversation mode is forced:

MIC -> LN

IR -> RECO

MICS, QR are "on"

4. APPLICATION COOKBOOK

In this chapter, the procedure for making a line-powered handsfree/handset basic application with the TEA1099 is given. With the help of fig. 64 in appendix, the design flow is given as a number of steps which should be made. As far as possible for every step, the components involved and their influence on every step are given.

Fig. 64 is the schematic of the OM5846 demoboard, so the values of the components are proposed but the adaptation to the application can be done by modifying these values.

Step	Adjustment
DC setting :	
Adjust the DC setting of the TEA1099 to the local PTT requirements.	
Voltage LN-GND	This voltage can be adjusted by increasing Vref up to 7 V at max line current with the Rva resistor between pins REG and SLPE.
DC slope	Not advised to modify.
Supply point VBB	Optimize the value of Cvbb.
Supply point VDD	Optimize the value of Cvdd.
Artificial inductor	Its value can be adjusted by changing the value of Creg: a smaller value speeds-up the DC current shape during transients but decreases the value of the inductance and therefore affects the BRL at low frequencies.
Impedance, sidetone and AGC :	
After setting the required set impedance, the sidetone has to be optimized using the sidetone network in order to minimize the loop gain in all line conditions. AGC can be adjusted at that step.	
Set passive impedance	The BRL is adjusted with the impedance network connected between LN and GND (Rcc + Rz//Cz in series with Ccz).
Sidetone	Adjust Zbal (Rbal1, Rbal2, Cbal) according to the line characteristics.
AGC	Internally defined, the characteristics (Istart and Istop) can be shiftted to higher line currents with an external Ragc resistor connected between AGC and LN. In case it is necessary to shift Istart and Istop to lower current values, the value of Rslpe may be slightly increased proportionally (see §3.1.4).
TEA1099 transmit and receive gains in handset mode	
Handset microphone gain	The microphone gain of the application has to be adjusted before entering pins MIC+/MIC- of the TEA1099. It can be reduced by using the resistor Rtx3 which forms a bridge attenuator with Rtx1 and Rtx2. Ctx1, Ctx2 form a high-pass filter with Rtx1, Rtx2 in series with the input impedance at MIC+/MIC-. A capacitor Cmich forms a low-pass filter with the impedance of the microphone and the resistors Rmicp/Rmicm. The gain between MIC+/MIC- and LN is 44.1 dB on 600 Ω.
Receive gain	Receive gain of the handset has to be adjusted with earphone amplifier with the resistor Re1. A capacitor Cgar in parallel with Re2 forms a low-pass filter. stability is ensured with capacitor Cgars (10 × Cgar) between pins GARX and VEE. The gain between IR and RECO is fixed at 29.7 dB.

Step	Adjustment
TEA1099 transmit gain in handsfree mode	
After the sensitivity and the curve of the microphone are adjusted, the gain can be adjusted to the desired value	
Microphone sensitivity	Rbmics sets the sensitivity and provides the polarisation of the electret.
Frequency curve	Cmicb with Rbmics and the output impedance of the electret form a low-pass filter.
Transmit gain and stability	Ctxin with the 20 kΩ input impedance at TXIN form a high-pass filter.
	Rgatx sets the microphone amplifier gain : $Gv(txin\text{-}txout)=20\times\log(0.7 \times Rgatx/Rstab)$
	The capacitor Catx in parallel with Rgatx forms a low-pass filter.
	The gain between HFTX and LN is fixed at 34.7 dB on 600 Ω.
	Chfx and HFTX input impedance form a high-pass filter.
	A resistor bridge attenuator may be inserted between TXOUT and HFTX or between Ctxin and TXIN.
TEA1099 loudspeaker amplifier :	
The gain is adjustable with Rgals, a high-pass filter can be made, the dynamic limiter timing can be chosen.	
Gain and frequency curve	Rgals sets the loudspeaker amplifier gain : $Gvlsao=20\times\log(0.35 \times Rgals/Rstab)$
	Cgals forms a low-pass filter with Rgals.
	Chrx and/or capacitor Clso in series with loudspeaker can form high-pass filters
Dynamic limiter timing	Capacitor Cdlc at pin DLC
TEA1099 receive channel	
The gain of the receive pass and the curve can be adjusted. The volume control range can be chosen.	
Receive gain	The gain between IR and RECO is 29.7 dB which compensate the sidetone attenuation minus 2.3 dB.
Receive curve	A bridge attenuator may be inserted between RECO and HFRX
	Chrx_with the input impedance of 20 kΩ at pin HFRX form a high-pass filter (a cut-off frequency between 100 and 200 Hz is advised),
Volume control	A linear potentiometer of 22 kΩ is suggested (3 dB for each 1.9 kΩ).

Step	Adjustment
TEA1099 Duplex controller (see § 3.5) :	
When all gains are adjusted, the switching range can be determined. Then the dial tone detector level followed by the sensitivities can be set. Finally the timings of the envelopes and the switching are adjusted.	
Switching range	Loop gain : Aloop=Atx1099 + Ast +Arx1099 + Aac-Asw < 0dB
	Choose Asw with safety margin
	Adjust Rswr : Asw=20log(Rswr/Rstab) with Rstab fixed at 3.65kΩ
Dial tone detector	Rrsen : Vdialtone = 2.5 µA×Rsen
Sensitivities	Rtsen for balanced sensitivities between Tx and Rx
	Ctsen form a high-pass filter with Rtsen
	Crsen form a high-pass filter with Rrsen
Signal envelopes	Ctenv (0.47 µF), Crenv (0.47 µF) :
	maximum attack : 140 µ/(3×Cenv) (dB/ms), release : 1 µ/(3×Cenv) (dB/ms)
Noise envelopes	Ctnoi (4.7 µF), Crnoi (4.7 µF) :
	attack : 1 µ/(3×Ctnoi) (dB/ms), maximum release : 140 µ/(3×Ctnoi) (dB/ms)
Switch-over timing	Cswt (220 nF) : δVswt/t=10µ/Cswt (mV/ms)
Ix-mode timing	Ridt (2.2 MΩ) : time constant=4×Ridt×Cswt

Fig. 55 Steps in the design flow of the TEA1099

5. APPLICATION EXAMPLES

In this chapter, some general block diagrams are provided to show the integration method of the TEA1099 in different terminal applications.

Moreover, a demoboard (OM5846) is available. As the TEA1099 may be used in various applications, this demoboard includes only the TEA1099 with its basic environment. Its schematic is shown in fig. 64 while its component placement diagram is in fig. 65.

On this schematic, the components which are connected with dotted lines are for RFI immunity purpose only. Moreover, a proposal of external antihowling circuitry is included on the layout of the PCB (its components are not equipped) with its input at TXOUT and output on DLC.

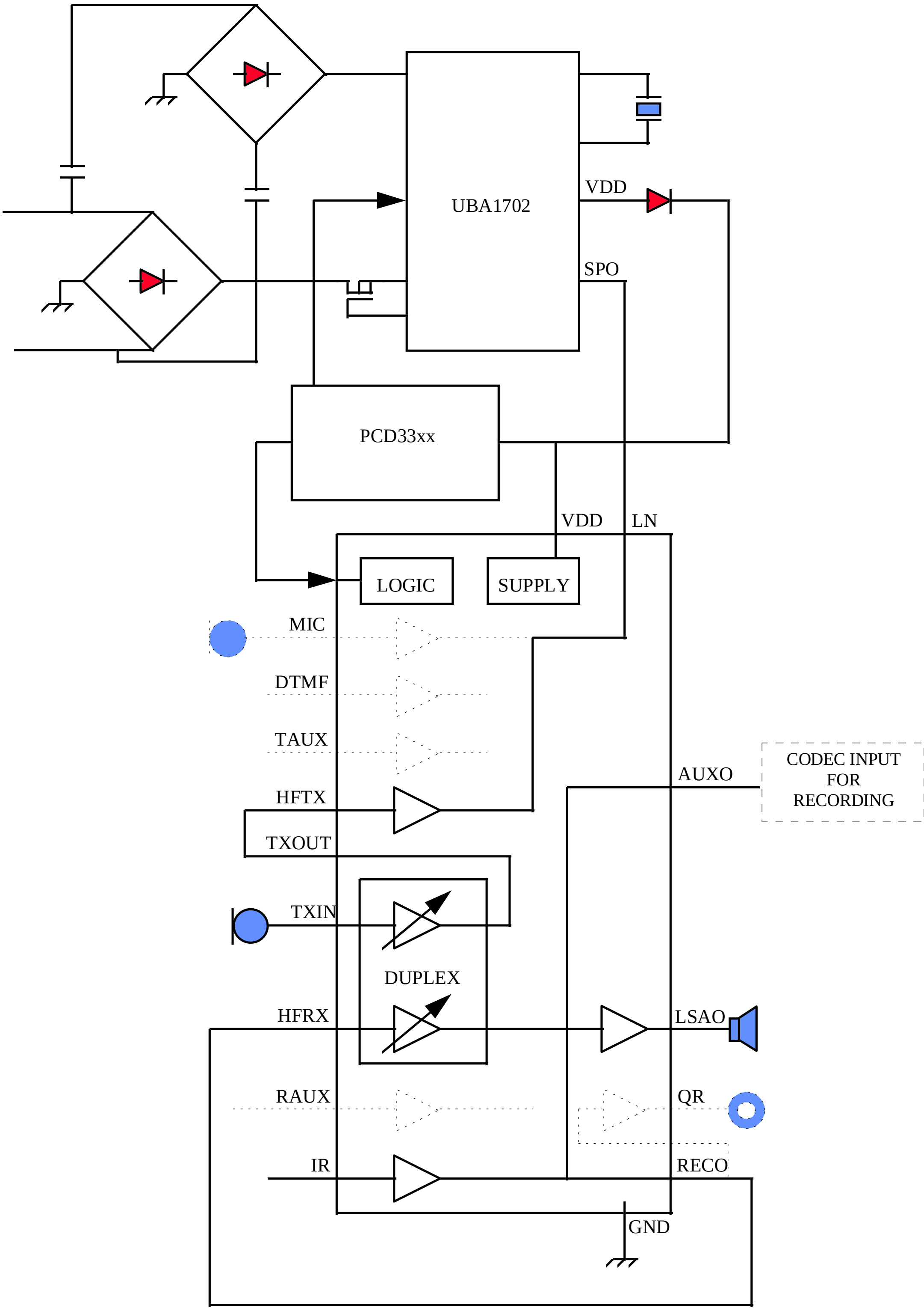

Fig. 56 Basic handsfree application

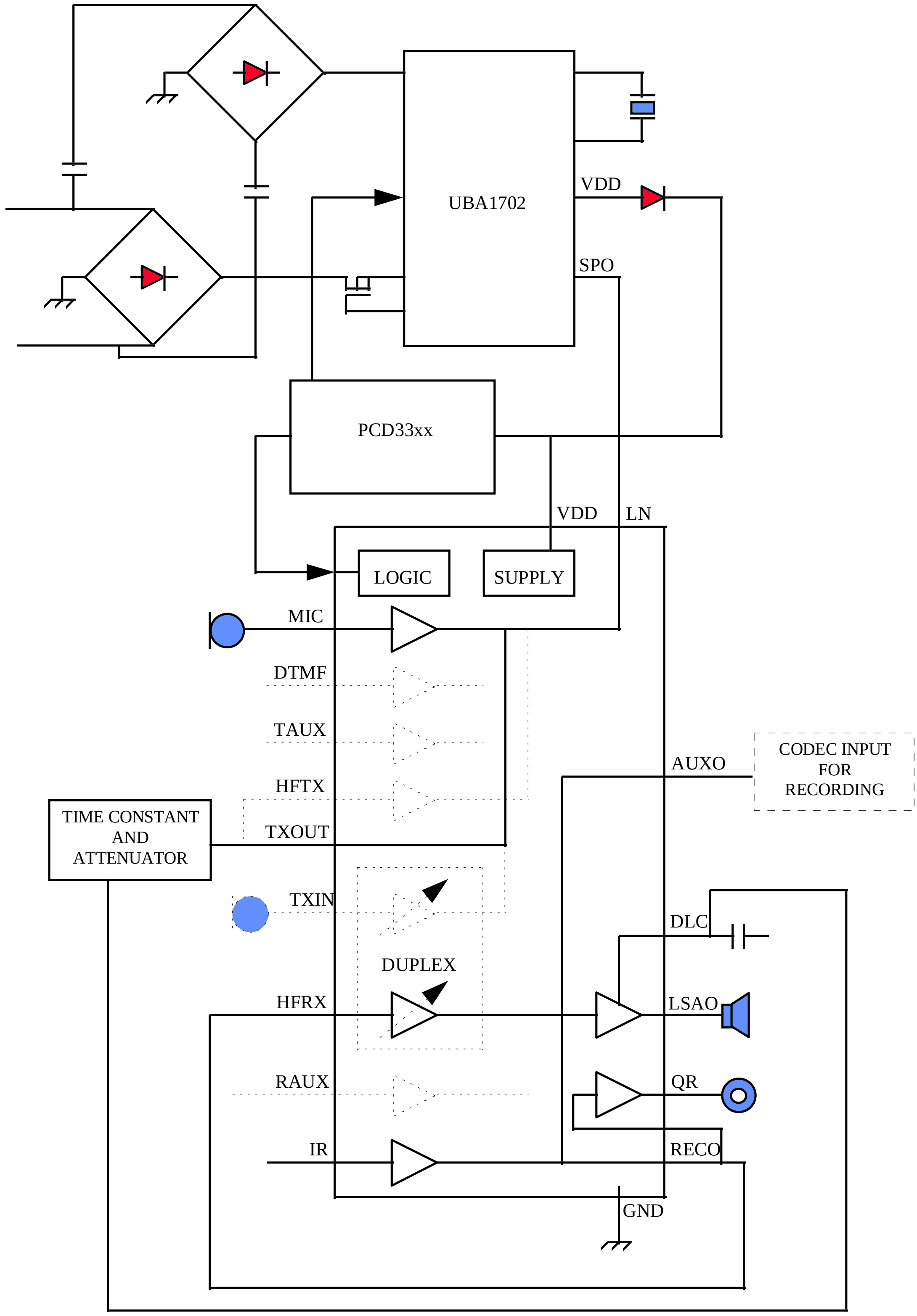

Fig. 57 Group-listenning conversation with antihowling

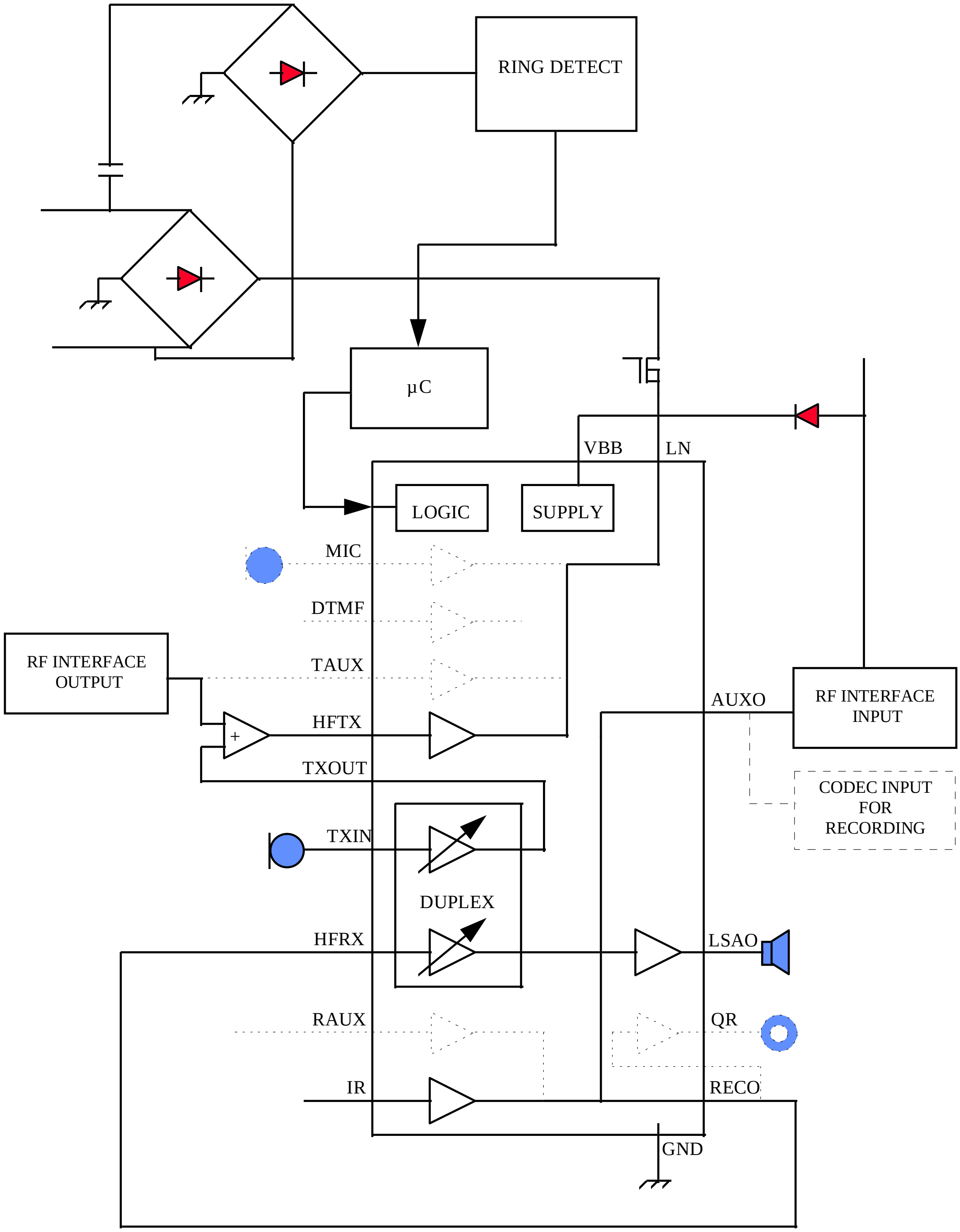

Fig. 58 Cordless: conference with line, base and mobile

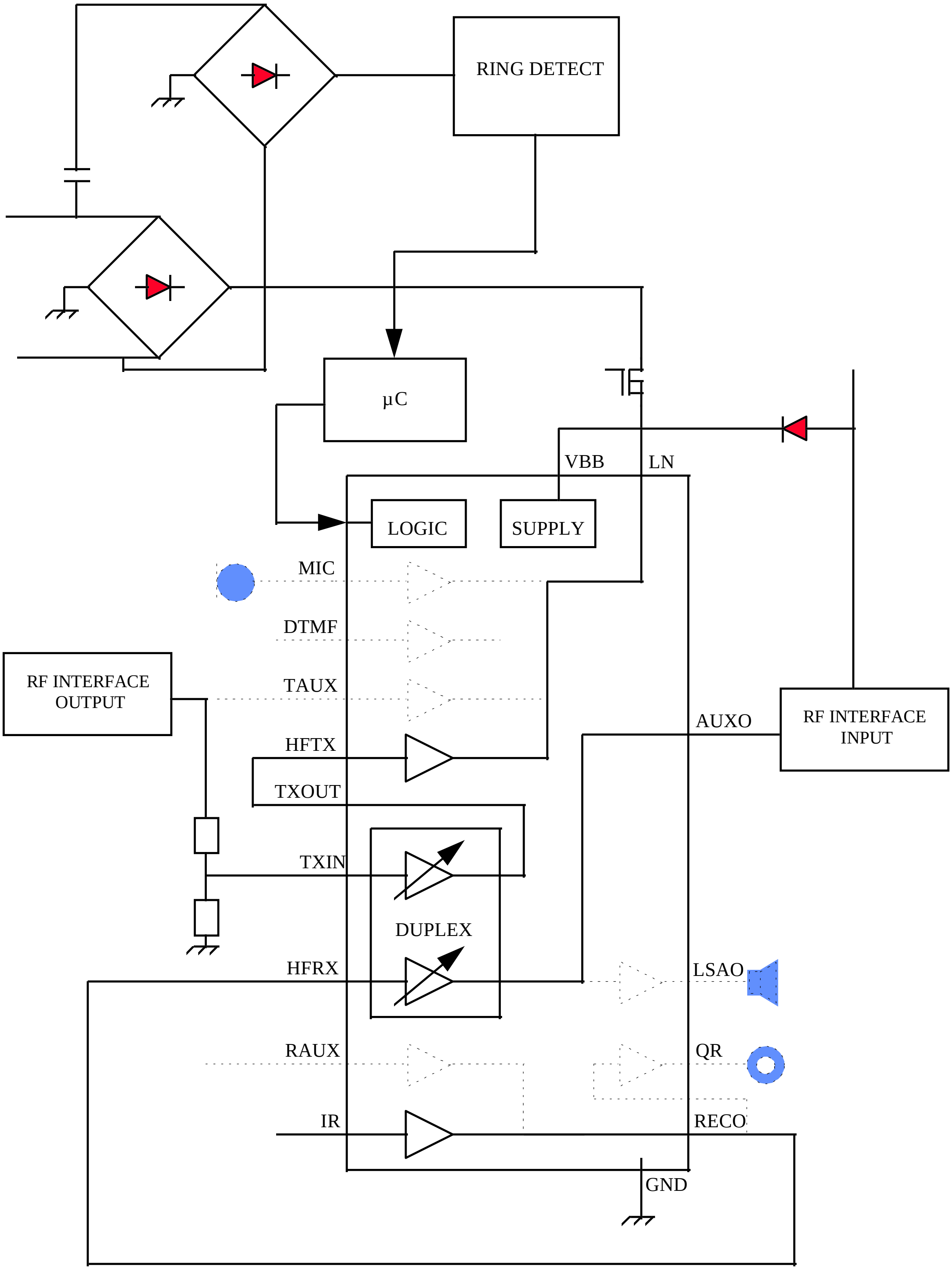

Fig. 59 Cordless: handsfree conversation in mobile

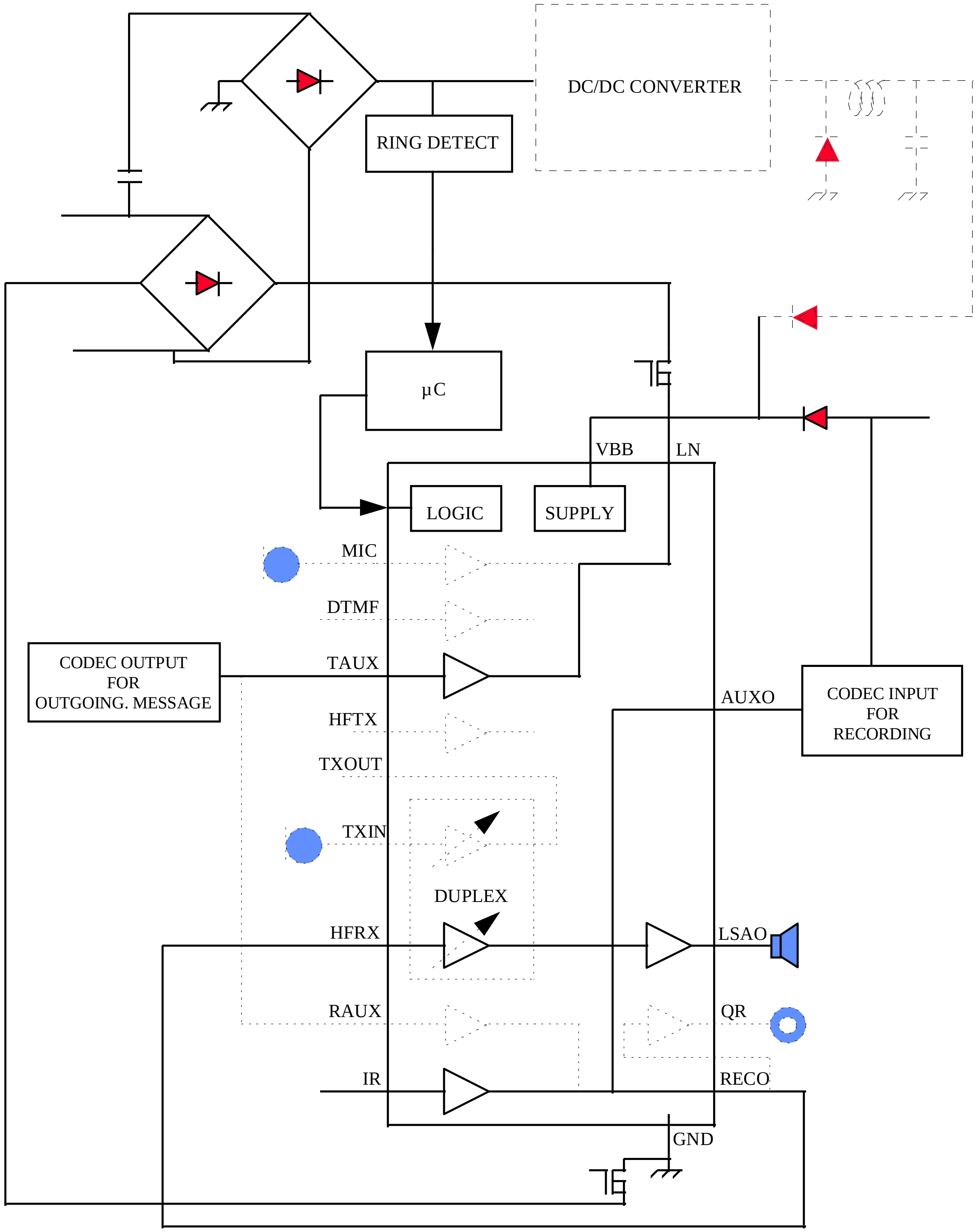

Fig. 60 Answering Machine on line

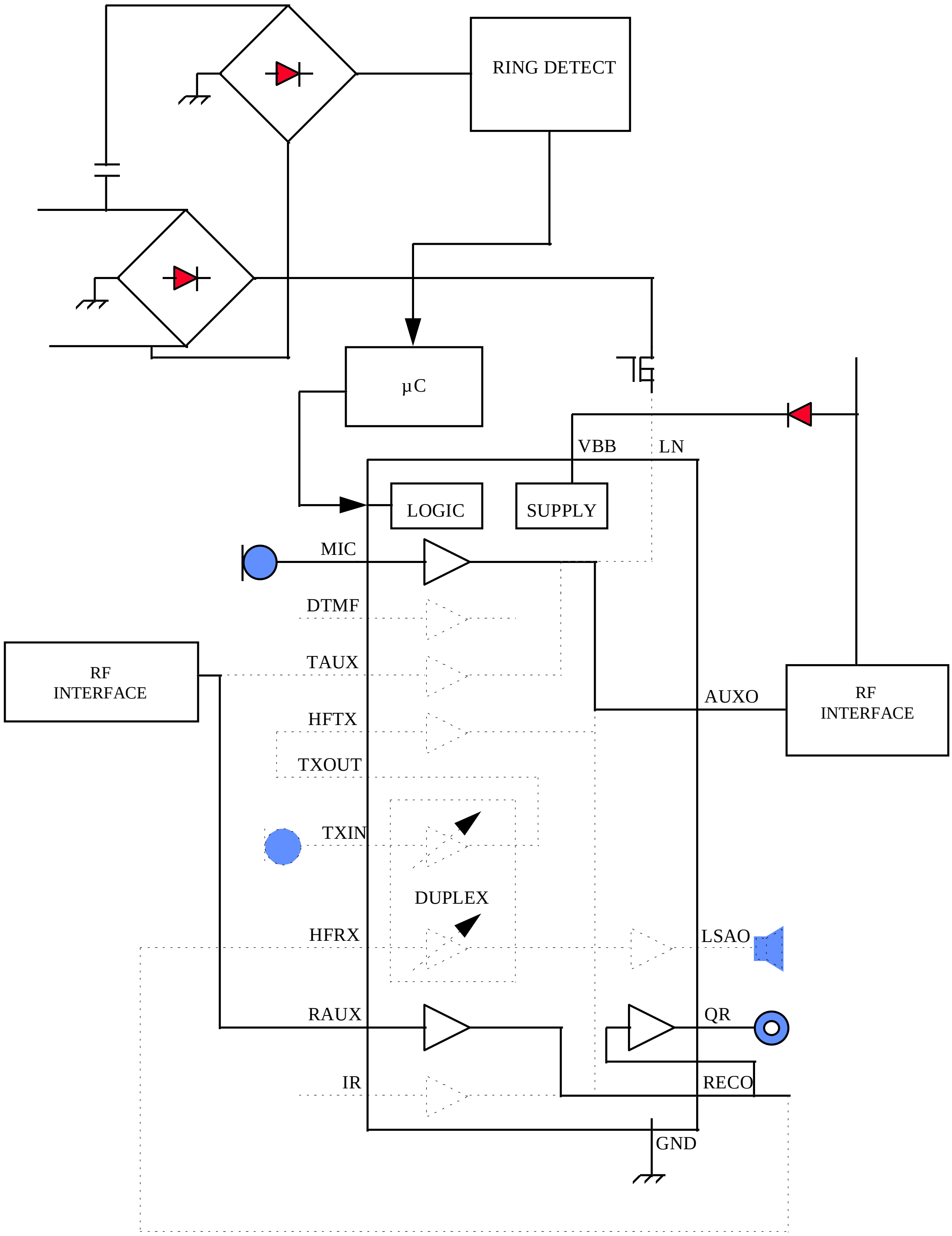

Fig. 61 Cordless intercom between mobile and base handset

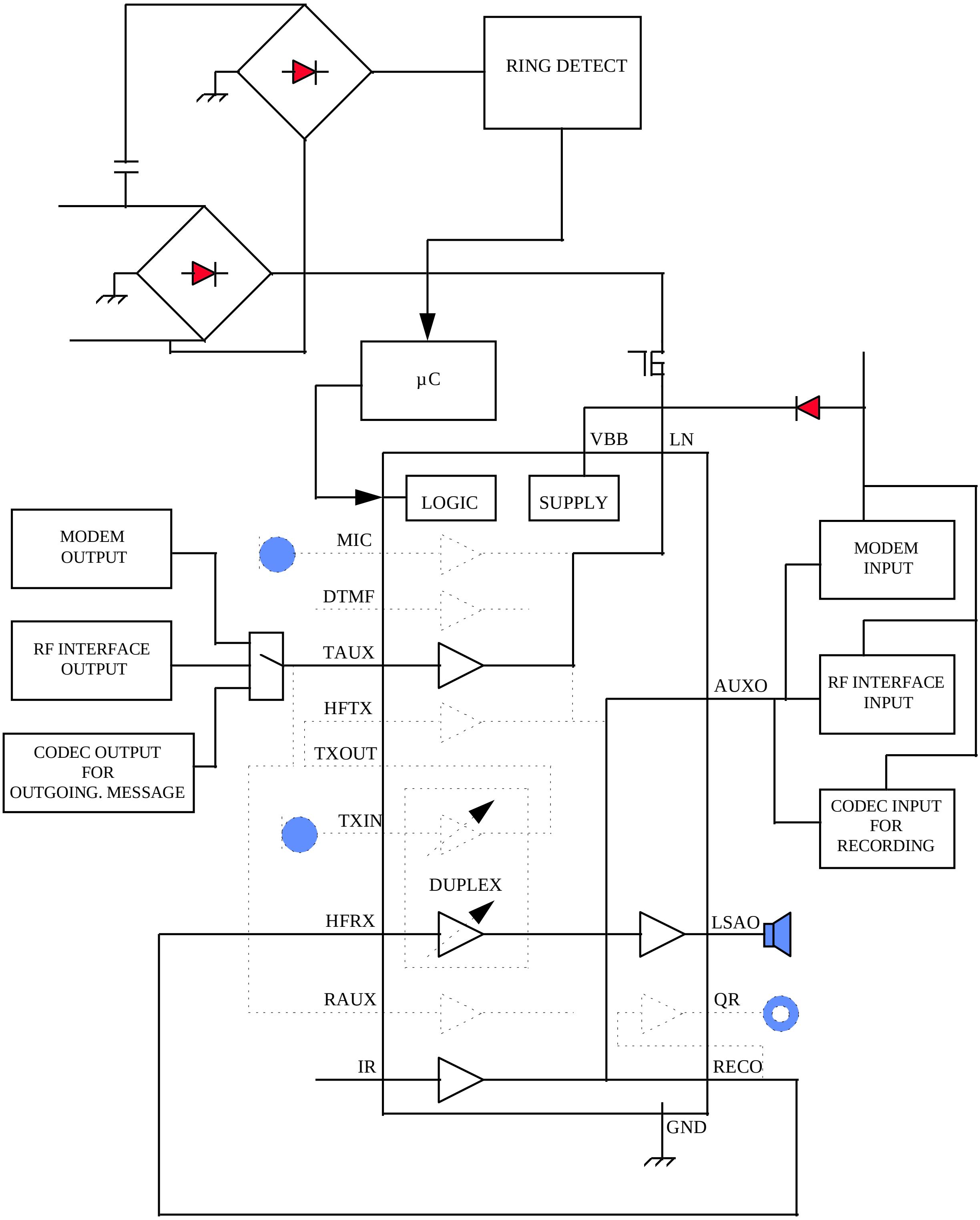

Fig. 62 Application with Fax, Cordless and Answering Machine

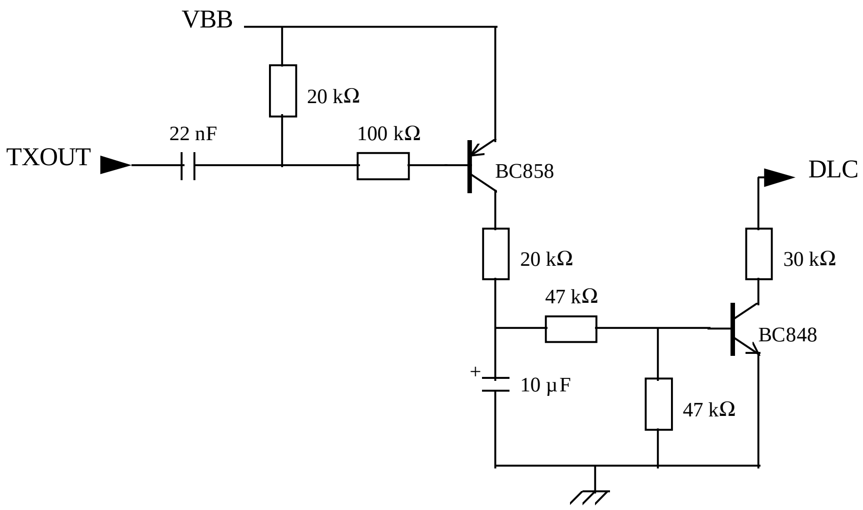

Fig. 63 Typical external antihowling circuit

6. ELECTROMAGNETIC COMPATIBILITY

As no common international specification exists for RFI immunity, and as different assembly methods may lead to different solutions, only some advices can be provided.

It is advisable to take care of the impedance of the GND, the smallest is always the best. Even if it is required to separate low level microphone signals on GNDTX from high level signals (loudspeaker or others), GND and GNDTX traces must be as wide as possible.

Also, the connection of Rstab, Rswr, Rgatx, Rgals and Rgarx has to be done with very short traces (specially STAB input which sets all the gains must be very immune).

VOL, MIC+/MIC-, HFTX, TXAUX and TXIN inputs may also be sensitive (RF signals entering these pins would be amplified). Rvol must preferably be connected with short traces or VOL input may be lightly decoupled by a capacitor to GND or better the trace must be inserted between GND traces. Care has to be taken with the lay-out of the microphone amplifiers, which is also helpfull for the noise, providing a good decoupling to GNDTX. Low-pass RC filters may be added at the inputs of the amplifiers (C3, C4, C8 on the demoboard). The output TXOUT may also be sensitive to high interference, it can be decoupled to GNDTX with a small capacitor (<56pF, Ctxor on the demoboard).

It can be helpfull to decouple the receive input IR, two possibilities are offered: a capacitor smaller than 220 pF between IR and GND (C10 on the demoboard) or a capacitor lower than 2.2 nF between IR and LN (C21 on the demoboard).

It is not allowed to put a capacitor directly between STAB and GND, only an RC network could be implemented if it helps (365 Ω, 4.7 nF).

Low impedance capacitors in parallel with the electrolythic one between VBB and GND may help.

Usually a low impedance capacitor connected between LN and GND helps for the conducted interferences, but this capacitor is in parallel with the impedance network of the apparatus, so, its value must be small enough.

In general when connections are coming from external environment (e.g. MIC+, MIC-, A, B on the demoboard), it is better to filter the RFI signal before it influences the close environment of the TEA1099 (e.g. action of C1,C2, C6, C11 which are close to the connectors on the demoboard).

7. REFERENCES

[1] TEA1099 Speech and Handsfree IC with auxiliary inputs/output and analog multiplexer

Device specification

[2] OM5846 Speech and Handsfree IC with auxiliary inputs/output and analog multiplexer Demonstration board

User Manual

[3] Philips Semiconductors

SEMICONDUCTORS FOR WIRED TELECOM SYSTEMS - IC03a -

Data handbook

[4] Philips Semiconductors

SEMICONDUCTORS FOR WIRED TELECOM SYSTEMS - IC03b -

Application handbook

APPENDIX

LIST OF ABBREVIATIONS AND DEFINITIONS

Aac	Electro-acoustic coupling (electrically measured)
AGC	Automatic line loss compensation of the TEA1099
Aloop	Loop gain of a handsfree telephone set
A.M.	Answering machine
Arx1099	Gain of the receive path of TEA1099
Ast	Sidetone gain
Asw	Switching range
Atsen	Gain from TXIN to TSEN of 40dB
Atx1099	Gain of the transmit path of TEA1099
AUXC	Logic input
AUXO	Auxiliary amplifier output of TEA1099
BRL	Balance Return Loss: matching between the apparatus impedance and a reference
Cgar	Capacitor setting receive path amplifier low-pass filter
Catx	Capacitor setting the base microphone amplifier low-pass filter
Cfeed	Microphone supply filter capacitor
Chrx	Receive input capacitor
Chfx	Transmit output capacitor
Clso	Loudspeaker coupling capacitor
Cmicb/h	Microphone low-pass filter capacitors
Crenv	Capacitor determining the receive signal envelope
Crnoi	Capacitor determining the receive noise envelope
Crsen	DC blocking capacitor of receive sensitivity setting
Cswt	Switch-over timing capacitor
Ctenv	Capacitor determining the transmit signal envelope
Ctnoi	Capacitor determining the transmit noise envelope
Ctsen	DC blocking capacitor of transmit sensitivity setting
Ctxin	Base microphone amplifier input capacitor
dBmp	dBm psophometrically weighted (0dBmp=1mW)
dBVp	dBV psophometrically weighted (0dBVp=1Vrms)

DTMF	Dual Tone Multi Frequency
δVswt	Voltage difference on SWT
GALS	Loudspeaker amplifier gain adjustment pin
GARX	Earphone amplifier gain adjustment pin
GATX	Base microphone gain adjustment pin
GL	Group-listening
GND	Ground reference pin
GNDTX	Ground reference pin for microphone signals
HF	Handsfree
HFC	Logic input (Handsfree on/off)
HFRX	Handsfree receive input
HFTX	Handsfree transmit input
IDT	Idle-mode timing adjustment pin
IR	Receive input from line
Istart	Start current of the AGC function
Istop	Stop current of the AGC function
Iswt	Output current through pin SWT (from decision logic)
Ix-mode	Idle mode
k	Scale factor of anti-sidetone network
Leq	Artificial inductor of the voltage stabilizer
LN	Positive line terminal of TEA1099
LSAO	Loudspeaker amplifier output of TEA1099
MIC+,MIC-	Microphone input of TEA1099
MICS	Microphone supply of TEA1099
MOSFET	Meta Oxide Field Effect Transistor
MUTT/	Logic input
MUTR/	Logic input
PCB	Printed circuit board
PD/	Logic input (power-down input)
PTAT	Proportional to absolute temperature
PTT	Public telephone company
QR	Earphone amplifier output of TEA1099
RAUX	Auxiliary receive input
RECO	Receive output from line

REG	Filter capacitor of the equivalent inductor connection pin of the TEA1099
RENV	Receive signal envelope timing adjustment pin
Rexch	Bridge resistance of exchange
RFI	Radio frequency interference
Rgarx	Resistor setting receive earphone amplifier gain
Rgatx	Resistor setting base microphone amplifier gain
Ridt	Resistor setting Ix-mode timing
Rload	Loudspeaker equivalent load resistor
Rmicm/p/b	Resistors setting microphone sensitivities
RNOI	Receive noise envelope timing adjustment pin
Rp	Internal resistance between LN and REG
Rrsen	Resistor setting sensitivity of the receive envelopes
RSEN	Receive signal envelope sensitivity adjustment pin
Rslpe	Resistor setting slope of the DC characteristic of TEA1096
Rstab	Resistor setting an internally used PTAT current
Rswr	Resistor setting switching range
Rtnoi	Resistor increasing microphone speech/noise threshold
Rtsen	Resistor setting sensitivity of the transmit envelopes
Rva	Voltage adjustment resistor
Rvol	Volume control potentiometer
Rx-mode	Receive mode
SLPE	DC slope pin of TEA1099
STAB	Reference current pin
SWR	Switching range adjustment pin
SWT	Switch-over timing adjustment pin
TENV	Transmit signal envelope timing adjustment pin
THD	Total Harmonic Distortion (%)
Tidt	Idle mode timing
TNOI	Transmit noise envelope timing adjustment pin
TSEN	Transmit signal envelope sensitivity adjustment pin
TXAUX	Auxiliary transmit input
TXIN	Base microphone amplifier input
TXOUT	Base microphone amplifier output
Tx-mode	Transmit mode

VBB	Positive supply of TEA1099
VDD	Microcontroller supply of TEA1099
Vdt, Vdialtone	Dial tone detector level
Vln	DC voltage between LN and GND
VOL	Volume adjustment pin
Vref	Stabilized reference voltage between LN and SLPE
Vslpe	DC voltage level between SLPE and LN
Zir	Input impedance of the receive amplifier of the TEA1099
Zbal	Anti-sidetone network
α	Gain control factor of the AGC

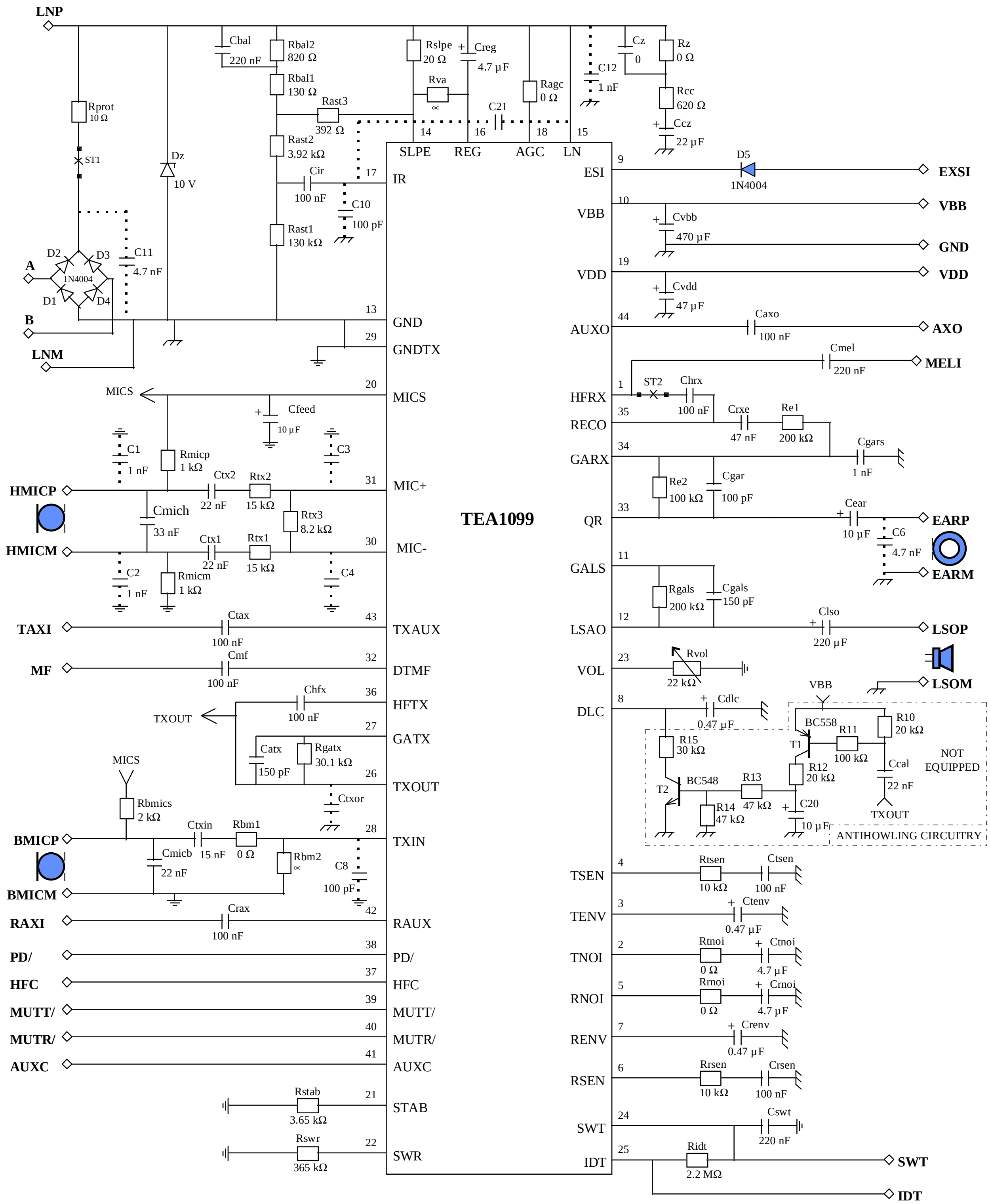

Fig. 64 Schematic of the demoboard

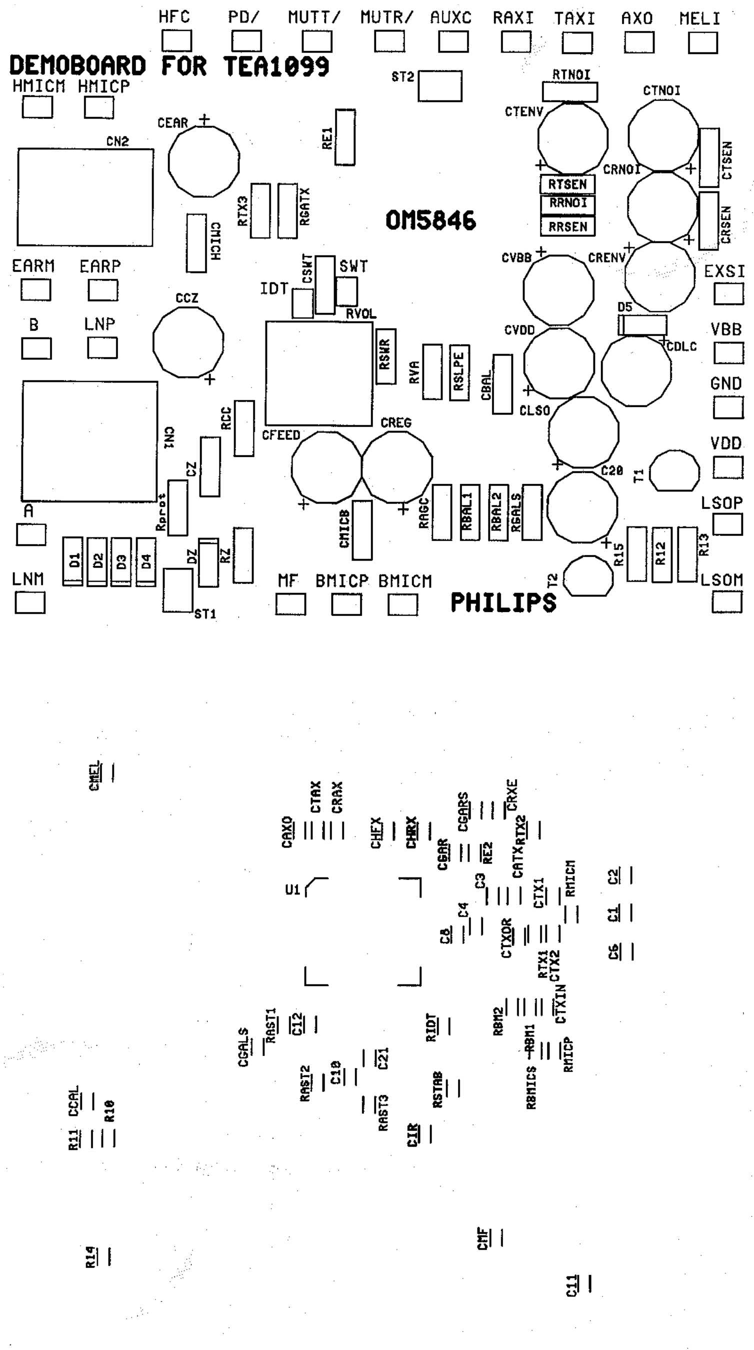

Fig. 65 component placement diagram of the demoboard

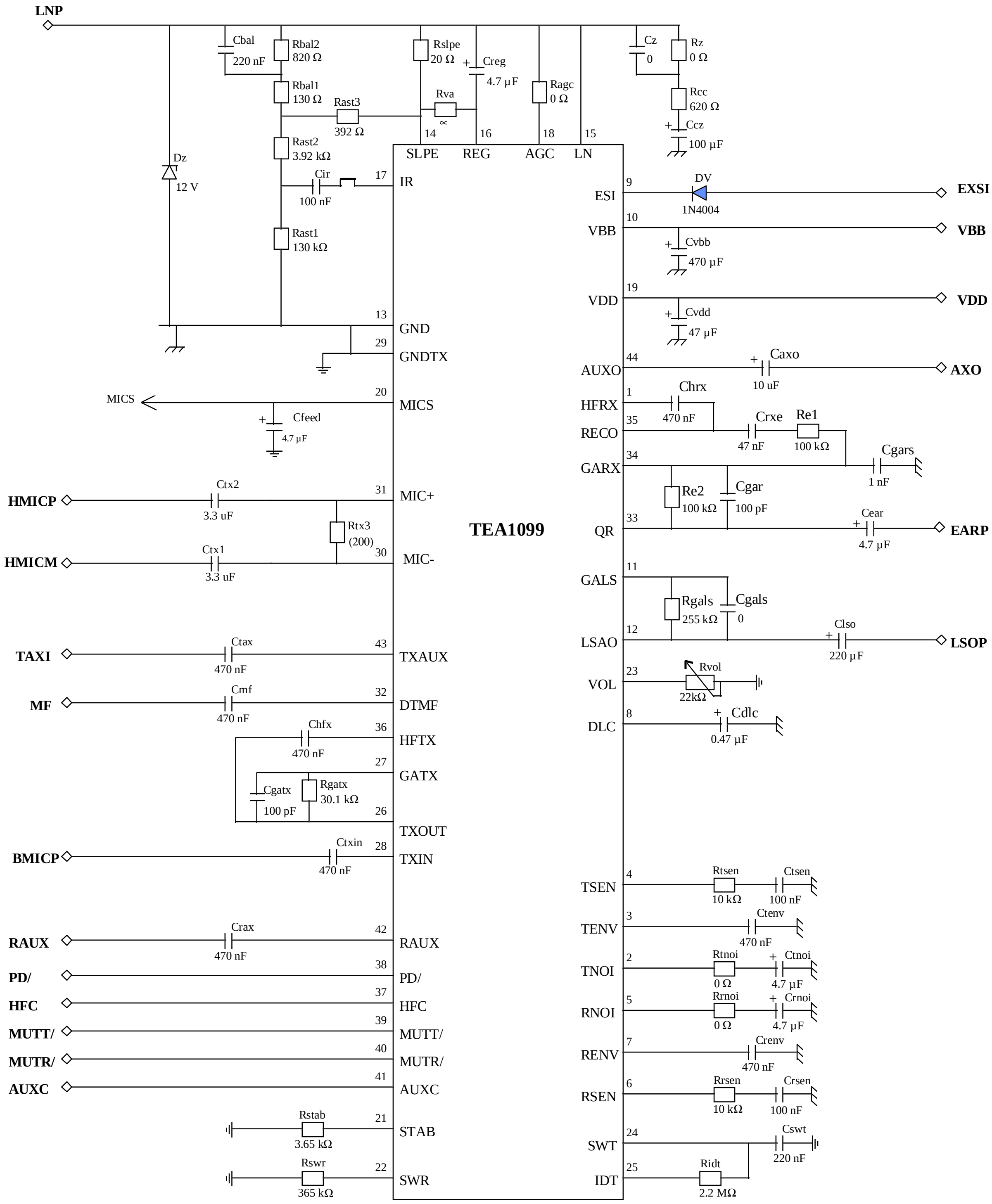

Fig. 66 Curve ref board of the TEA1099