## APPLICATION NOTE

# TEA1095
# VOICE SWITCHED
# SPEAKERPHONE IC

**CTT/AN95083**

**Author:**
**Jean-Marie Malaurie**
**Technical Marketing, Telecom Products**
**Caen, France**

**Keywords**

Handsfree
TEA1095
TEA1096
Duplex controller
Transmit channel /path
Receive channel /path

**Date: September 1st, 1995**

**Philips**
**Semiconductors**

PHILIPS

**Abstract**

*The TEA1095 is a high performance, low power consumption voice switched speakerphone IC designed for integration of a handsfree function in terminal environments.*

*A detailed description of the IC, advices on adjustments and a worked-out application example are contained in this report.*

**Summary**

A detailed description of the TEA1095 is given. In conjunction with TEA1096 ( transmission circuit with built-in loudspeaker amplifier ), it offers a handsfree function. The TEA1095 incorporates a microphone amplifier and a duplex controller with signal and noise monitors on the transmit and receive channels. A cookbook gives the general application steps.

**CONTENTS**

## 1. INTRODUCTION

The TEA1095 is a circuit which, in combination with TEA1096 ( transmission circuit with built-in loudspeaker amplifier ), offers a handsfree function. It incorporates a microphone amplifier, a volume control of the receive channel and a duplex controller with signal and noise monitor on the transmit and receive channel. In contrary with the Philips handsfree circuits TEA1093 and TEA1094, the TEA1095 has neither integrated supply nor loudspeaker amplifier. This makes the TEA1095 more flexible for implementation in applications with external loudspeaker amplifier, external supply, such as cordless telephones and answering machines.

The function of the handsfree application will be illustrated with the help of fig 1.1.

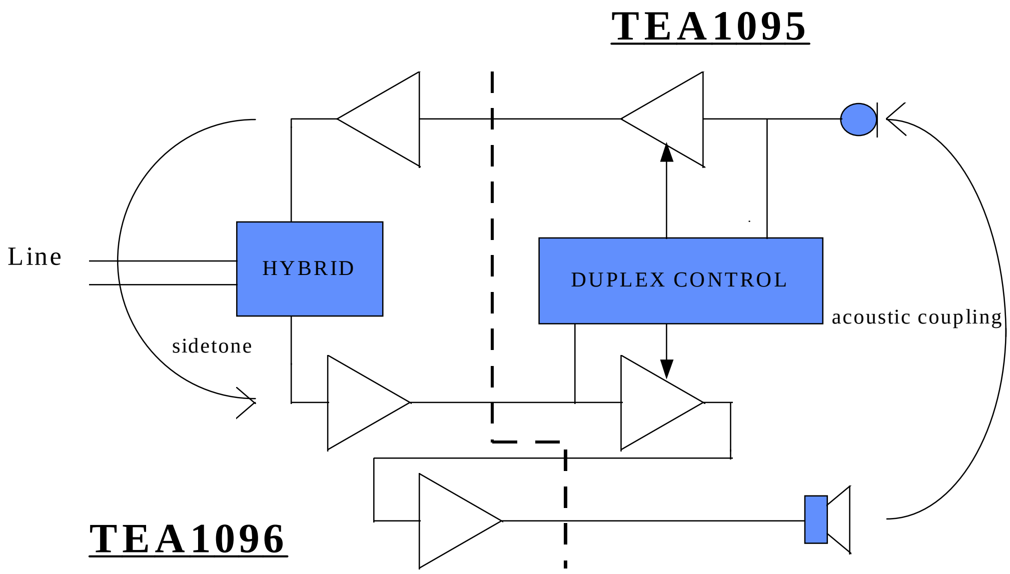


Fig.1.1 Handsfree telephone set principle

The left side of fig 1.1 shows a principle diagram of a part of the TEA1096 circuit by means of a receiving amplifier, a transmit amplifier, the loudspeaker amplifier and the hybrid. The right side of fig 1.1 shows a principle diagram of a part of the TEA1095 circuit by means of the microphone amplifier and the duplex controller.

As can be seen from fig 1.1, 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 TEA1095 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. This is done such that 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 microphone amplifier 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 gain of the microphone amplifier 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 TEA1095 and its application with the TEA1096. The description is given by means of the block diagram of the TEA1095 (&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 appendices contain a measurement setup for the electro-acoustical adjustment of the TEA1095 handsfree application (A), a list of abbreviations (B) and application diagrams of the TEA1095 (fig. C1, C2).

## 2. BLOCK DIAGRAM

In this chapter, the block diagram of the TEA1095 is shown by means of fig.2.1. The pinning of the TEA1095 is given by means of fig.2.2. A short description of the block diagram is given including the function of the external components.

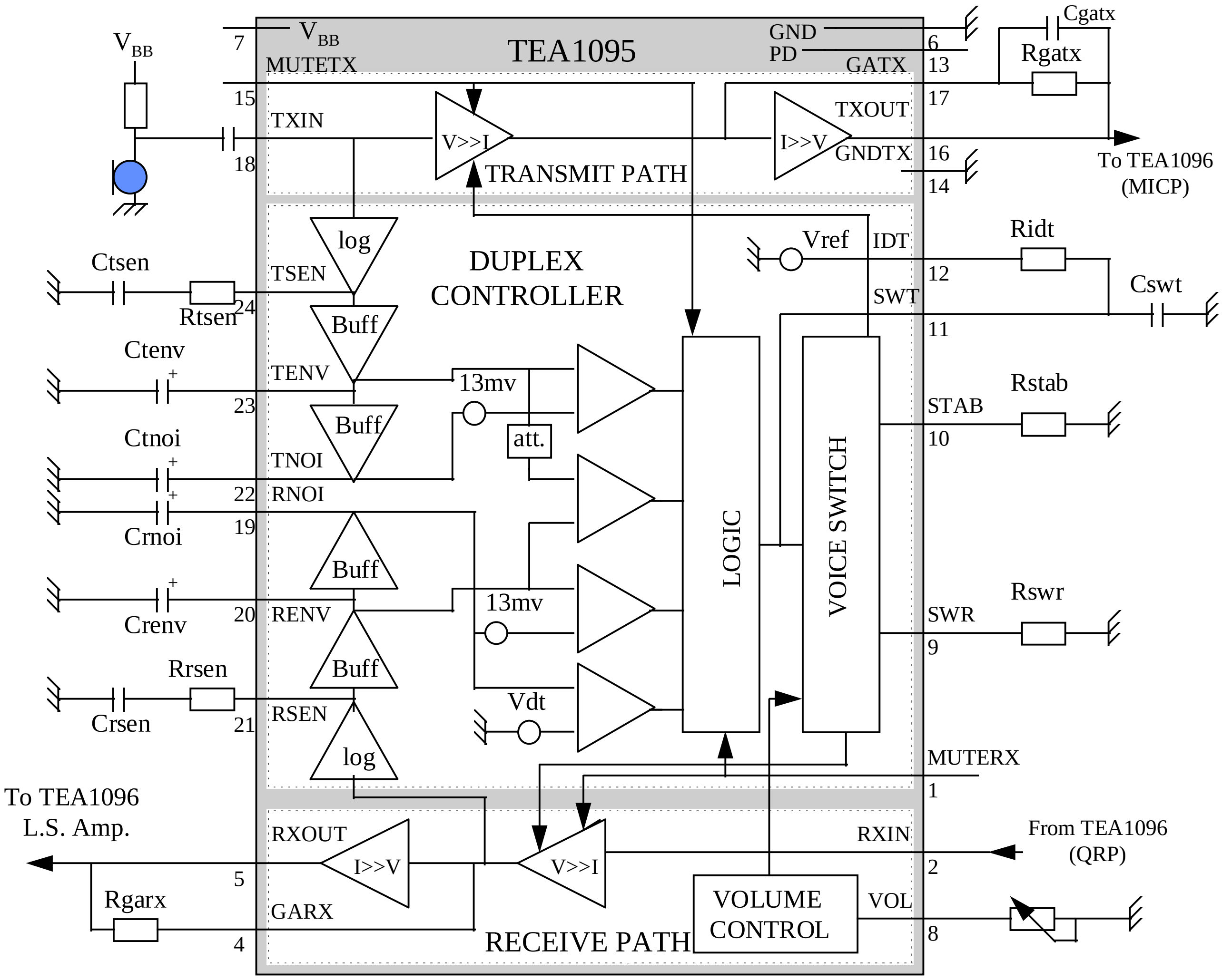


Fig. 2.1 Block diagram of TEA1095

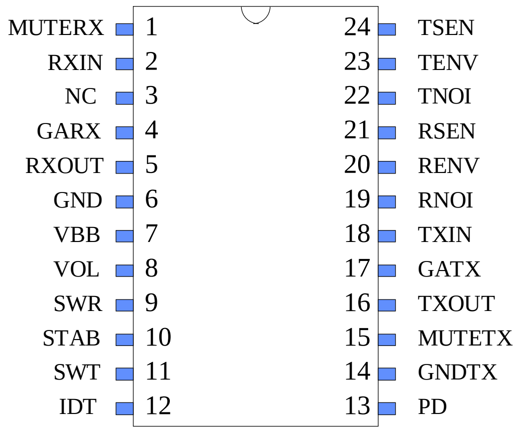


Fig. 2.2 Pinning of TEA1095

| PIN | NAME | DESCRIPTION |
|---|---|---|
| 1 | MUTERX | Receive channel mute input |
| 2 | RXIN | Receive channel input |
| 3 | NC | Not connected |
| 4 | GARX | Receive gain adjustment |
| 5 | RXOUT | Receive channel output |
| 6 | GND | Ground reference |
| 7 | VBB | Positive supply input |
| 8 | VOL | Receive channel volume adjustment |
| 9 | SWR | Switching range adjustment |
| 10 | STAB | Reference current adjustment |
| 11 | SWT | Switching timing adjustment |
| 12 | IDT | Idle-mode timing adjustment |
| 13 | PD | Power-down input |
| 14 | GNDTX | Ground reference for microphone amplifier |
| 15 | MUTETX | Transmit channel mute input |
| 16 | TXOUT | Transmit channel output |
| 17 | GATX | Microphone gain adjustment |
| 18 | TXIN | Microphone amplifier input |
| 19 | RNOI | Receive noise envelope timing adjustment |
| 20 | RENV | Receive signal envelope timing adjustment |
| 21 | RSEN | Receive signal envelope sensitivity adjustment |
| 22 | TNOI | Transmit noise envelope timing adjustment |
| 23 | TENV | Transmit signal envelope timing adjustment |
| 24 | TSEN | Transmit signal envelope sensitivity adjustment |

In fig.2.1 it can be seen that the IC consists out of four parts: the supply, the microphone amplifier, the receive path and the duplex controller. These blocks will be shortly described below including the function of the external components. The detailed description will follow in chapter 3.

**Supply :**

The circuit is supplied between pins VBB and GND. The TEA1095 can be switched into a low power consumption mode with the pin PD.

**Microphone amplifier :**

The handsfree microphone signal is amplified from pin TXIN to pin TXOUT. The signal reference is GNDTX, a “clean ground” which has to be connected to GND. The input TXIN has to be coupled to the microphone by means of a capacitor. The gain of the amplifier can be set with Rgatx. This amplifier can be muted by making pin MUTETX high.

**Receive path :**

The receive signal is amplified from pin RXIN to pin RXOUT. The input RXIN has to be coupled by means of a capacitor. The gain of the amplifier can be set with Rgarx, and the volume of the receive signal can be adjusted by means of the potentiometer connected between input VOL and GND. This channel can be muted by making pin MUTERX high.

**Duplex controller :**

From both the transmit and receive signal, 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 logic determines to 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.

# 3. DESCRIPTION OF THE TEA1095

This chapter describes in detail the four blocks of the duplex controller circuit TEA1095: the supply (3.1), the microphone amplifier (3.2), the receive path (3.3) and the duplex controller (3.4). 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 of the TEA1095 specification, see TEA1095 device specification.

## 3.1 Supply block

**Principle of operation**

As opposite to TEA1093 which includes an integrated supply which stabilizes a supply voltage out of the line current and as the TEA1094, the TEA1095 has no integrated supply. This makes the TEA1095 most suitable for applications where the handsfree controller is supplied from an external voltage source.

In fig. 3.1.1, different supply arrangements with TEA1095 are shown.

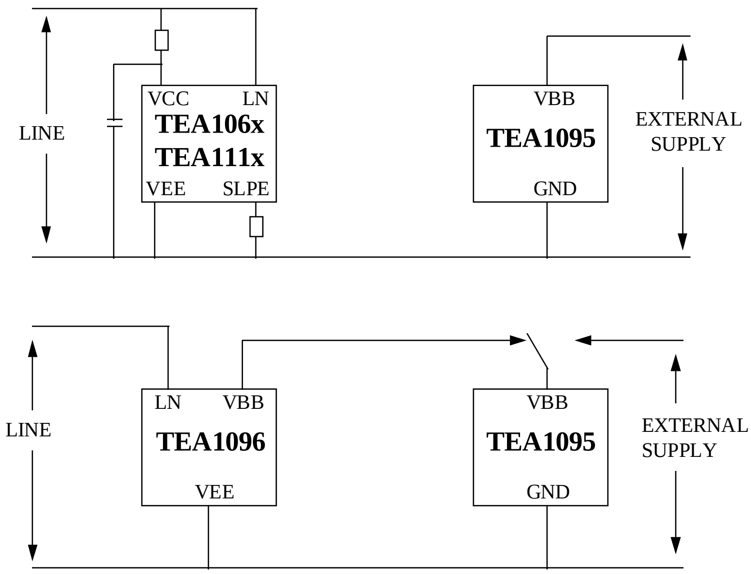


Fig. 3.1.1 Supply arrangement examples with TEA1095

As can be seen, the line current flows through the TEA106x or the TEA1096, while the TEA1095 is supplied from an external voltage source. Nevertheless, as can be seen in the bottom example, TEA1096 is able to provide this supply, leading to a line-powered very integrated solution. The common reference for all signals is GND on TEA1095 and VEE on TEA106x or TEA1096 side.

In case of a line-powered application using TEA106x or TEA111x, the TEA1095 as well as an external loudspeaker amplifier can be supplied via a large coil (or TEA1081) connected between line and VBB. At VBB,

a capacitor has to be connected to serve as reservoir. The TEA1095 and the loudspeaker amplifier have to be referenced at SLPE in order not to influence the transmission characteristics.
In fig.3.1.1 no galvanic insulation is drawn. When such is needed, the insulation can be done in the supply part or in the signal interfacing. In the supply part, the galvanic insulation can be done in the main-adaptor. In the signal interfacing between the TEA1095 and the TEA106x, the galvanic insulation can be done either with transformers or with opto-couplers.
The power consumption of the TEA1095 can be dramatically reduced when its functionalities are not required by making pin PD high.

**Adjustments and performances**

The voltage which can be applied between VBB and GND may vary between 2.9 V and 12 V. The 12 V is an absolute maximun rating. When a supply is used which may generate, during transients, a higher voltage, an appropriate protection device must be applied between VBB and GND in order to control this voltage.
The current consumption of the TEA1095 is typically 2.7 mA at VBB = 5 V. At higher voltages, the current consumption slightly increases.
In power-down mode, the current consumption is typically reduced to 140 µA at VBB = 5 V.

## 3.2 Microphone amplifier block

In the first paragraph of this chapter, the principle of operation of the microphone amplifier is described as well as its adjustments and performance. In the second paragraph, the mute transmit function is described.

### 3.2.1 Microphone amplifier

**Principle of operation**
In fig. 3.2.1 the block diagram of the microphone amplifier of the TEA1095 is depicted together with the interconnection with the TEA1096.

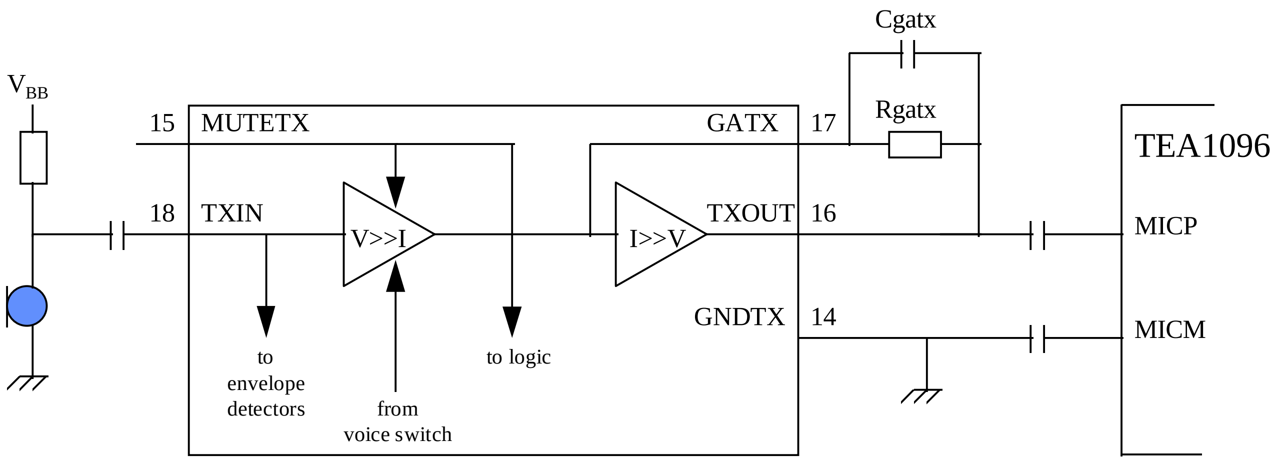


Fig. 3.2.1 Block diagram of the microphone path

As can be seen in fig.3.2.1, the microphone amplifier is referenced to pin GNDTX instead of being referenced to GND. This is in order to prevent interference from other blocks of the TEA1095 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 referenced to GND.

The input of the microphone amplifier is pin TXIN. It is an assymmetrical input well suited for electret microphones. Induced signals in the short wire between the microphone and pin TXIN are assumed to be negligible. This in contrary with the handset microphone which is connected via the handset cord. The TEA106x/TEA111x families as well as the TEA1096 have symmetrical microphone inputs.

The output of the microphone amplifier is pin TXOUT. When interconnecting the TEA1095 and the TEA1096, pin TXOUT is preferably connected to the TEA1096 input MICP, in that case, pin MICM of the TEA1096 is connected to pin GNDTX.

As can be seen in fig.3.2.1, the 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 & 3.4. The gain of the end-amplifier is determined by the external feedback resistor Rgatx.

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

Atx = 20 * log ( 0.72 * Rgatx / Rstab ).

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

**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 of 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 VBB via a resistor. However, during normal operation, VBB may contain a small riple, and due to poor power supply rejection of electret microphones it is advised to add an RC smoothing filter in the feeding part, as shown in fig.3.2.2.

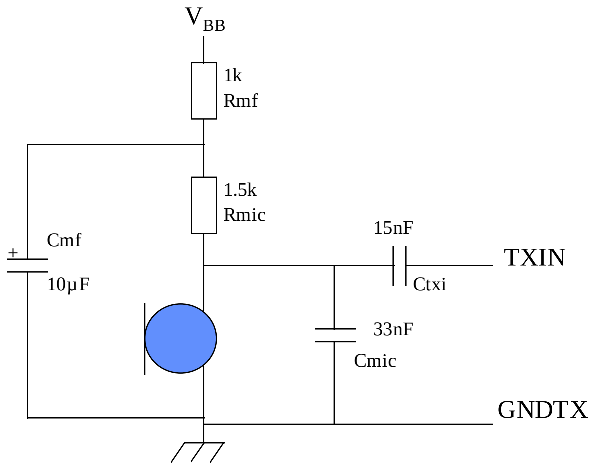


Fig. 3.2.2 Supply arrangement for electret microphone

As shown in fig.3.2.2, the RC smoothing filter is referenced to GNDTX in order to have one good reference for the whole microphone signal path. On the printed circuit board lay-out GNDTX can be connected with a separate wire to pin VEE of the TEA1096 in order to reduce ground interference as much as possible.

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 -15 to +25 dB to suit application specific requirements. With the resistor Rgatx = 30.1 kΩ, the gain equals typically 15.5 dB.

Capacitor Cgatx is applied in parallel with resistor Rgatx to ensure stability of the microphone amplifier, it also provides 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, see &3.4. At 10 mVpeak 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 output drive capability at pin TXOUT is 20 µArms.

The output noise at TXOUT of the TEA1095 is -100 dBmp (psophometrically weighted) at a gain of 15 dB. With a sending gain of the TEA1096 set at 35 dB (total handsfree transmit gain of 50 dB), the noise level on the line will be :  -65 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 -110 dBmp.

The bottom level of the sending noise on the line is determined either by the speech circuit (TEA1096 or TEA106x/TEA111x) or by the noise at TXOUT increased by the transmit gain of the speech circuit, whichever is largest. When in Ix-mode, the noise on the line is due to TXOUT noise level, it can be reduced by changing the distribution of the gains between TEA1095 and speech circuit : increasing the transmit gain of the TEA1095 and decreasing with the same value the gain on the speech circuit side by modification of the attenuation placed between TXOUT and MICP of the speech circuit.

### 3.2.2 Transmit mute

During handsfree operation, the microphone can be muted by making pin MUTETX high, so conversation cannot be heard by the other party. When the microphone amplifier is muted, automatically the TEA1095 switches over to the RX-mode, see also &3.4.

When a logic high is applied to MUTETX, meaning the voltage on MUTETX is higher than 1.5 V, the microphone preamplifier is muted. The end-amplifier can still be used by applying a current signal on GATX. The obtained gain reduction is 80 dB. The current which has to be sourced into pin MUTETX when high is typically 2.5 µA.

When MUTETX is logic low, meaning the voltage on MUTETX is lower than 0.3 V or pin MUTETX is left open, the microphone amplifier is not muted.

The maximum allowable voltage on pin MUTETX is VBB+0.4 V, the minimum allowable is GND-0.4 V.

## 3.3 Receive path block

The block diagram of the complete receive path block is depicted in fig.3.3.1

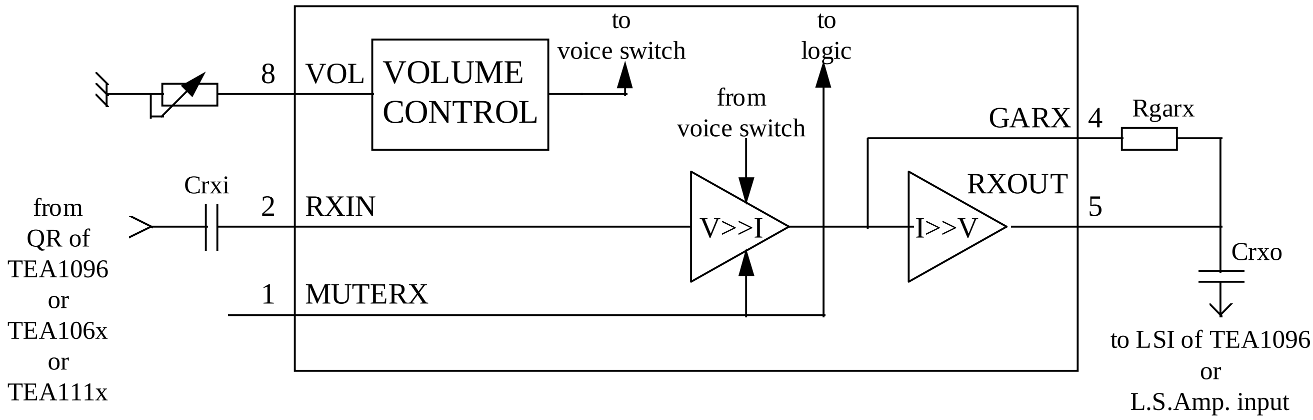


Fig.3.3.1 Principle of the receive path

As can be seen in fig.3.3.1, the receive channel is built up out of two parts: the receive path itself and the volume control. In the first paragraph of this chapter, the principle of operation of the receive path is described as well as its adjustments and performances. In the second paragraph, the same items are described for the volume control. In the third paragraph, the receive mute function is described.

### 3.3.1 Receive path

**Principle of operation**

As can be seen in fig.3.3.1, the input of the receive channel, pin RXIN is assymetrical and the signal has to be referenced to GND. The input RXIN can be connected, via a decoupling capacitor, to the earphone output QRP of the speech circuit (TEA1096 or TEA106x/TEA111x).

The output of the receive channel is pin RXOUT. When interconnecting the TEA1095 and the TEA1096, pin RXOUT is connected via a decoupling capacitor to input pin LSI of the TEA1096.

As can be seen in fig.3.3.1, the receive path itself 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 Rgarx.

The overall gain (Arx) of the receive path from input RXIN to output RXOUT is given as:

$$Arx = 20 * \log( 0.46 * Rgarx / Rstab ).$$

With Rstab being the resistor at STAB of 3.65 kΩ.

**Adjustments and performances**

The input signal for the receive channel has to be coupled in via the capacitor Crxi to block DC. Together with the input impedance of 20 kΩ at RXIN, 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 RXIN can handle signal up to 390 mVrms with a total harmonic distortion of 2%.

The input RXIN is biased at around 0V with respect to GND. By applying a signal to the input, it can become negative. The protection on this pin is made different from other pins which makes it possible to have RXIN as low as -1.2 V without damaging the circuit.

The output RXOUTcan be connected to the LSI input of TEA1096 or the input of any loudspeaker amplifier via a decoupling capacitor Crxo. The output is biased at 1.4 V referenced to GND. Together with the input impedance of the loudspeaker amplifier, Crxo forms a first order high-pass filter. With the resistor Rgarx, the gain of the receive path can be adjusted from -20 to +20 dB. The gain equals typically 6.3 dB with resistor Rgarx=16.2 kΩ. A capacitor Cgarx can be connected in parallel with Rgarx to provide a low pass filter which can be used to adjust the loudspeaker amplifier curve.

The output drive capability at pin RXOUT is 100µArms.

The noise level at the output RXOUT is -91 dBmp at a gain of 6 dB and with the input RXIN shorted with 200 Ω to GND.

### 3.3.2 Volume control

**Principle of operation**

Via the volume control block, 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.

**Adjustments and performances**

Out of pin VOL a current Ivol, set by Rstab, see &3.4, is flowing which is proportional to the absolute temperature (PTAT). At room temperature this current is around 10 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 950 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 10 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 &3.4. 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 saturation voltage. Due to saturation voltage, it is advised not to use bipolar transistors as switches.

When a voltage is applied to pin VOL to control the volume, preferably this voltage has to be a PTAT voltage source. If not, the obtained gain varation is no longer temperature compensated.

### 3.3.3 Receive mute

During handsfree operation, the receive channel can be muted by making pin MUTERX high. As a result the receive signal is reduced by 80 dB, also the TEA1095 is internally forced into Tx-mode.

When a logic high is applied to MUTERX, meaning a voltage higher than 1.5 V, the receive preamplifier is muted. The end amplifier can still be used by applying a current signal on GARX. The obtained gain reduction is 80 dB when the volume control is set at maximum volume. The current which has to be sourced into pin MUTERX when high is typically 2.5 µA.

When MUTERX is logic low, meaning the voltage is lower than 0.3 V or pin MUTERX is left open, the receive path is muted.

The maximum allowable voltage on pin MUTERX is VBB+0.4 V, the minimum allowable is GND-0.4 V.

## 3.4 Duplex controller block

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.3.4.1.

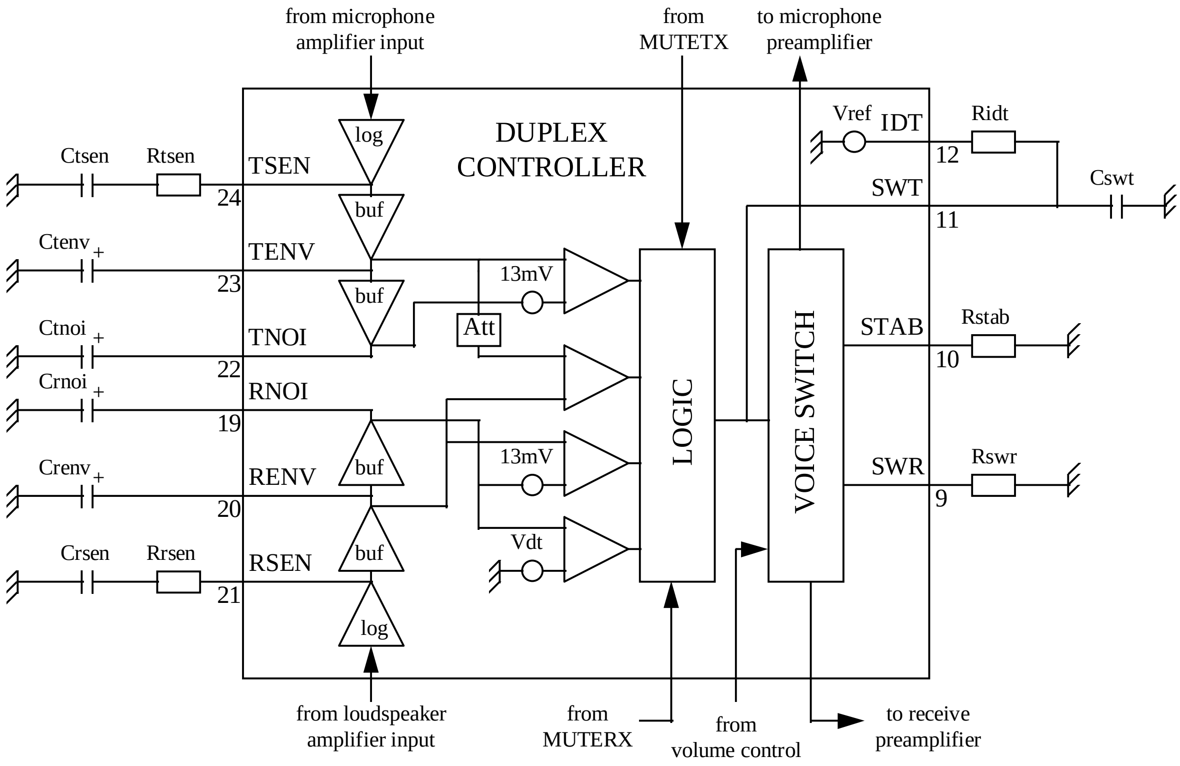


Fig. 3.4.1 Principle of the duplex controller

As can be seen in fig.3.4.1, the duplex controller is built up out of signal and noise envelope detector, decision logic and a voice switch.

The signal and noise envelope detectors determine the signal envelope and the noise envelope of both the transmit and receive signal. These envelopes are used by the decision logic to determine to which mode the TEA1095 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 TEA1095 between the three modes while keeping the loopgain constant.

In paragraphs 3.4.1 to 3.4.3, the principle of operation of the three parts is given. In paragraph 3.4.4, the adjustments and performances of the complete duplex controller are given.

### 3.4.1 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.3.4.1. 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.3.4.1, 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.3.4.2 where the signal and noise envelope of one channel are depicted together with the input signal.

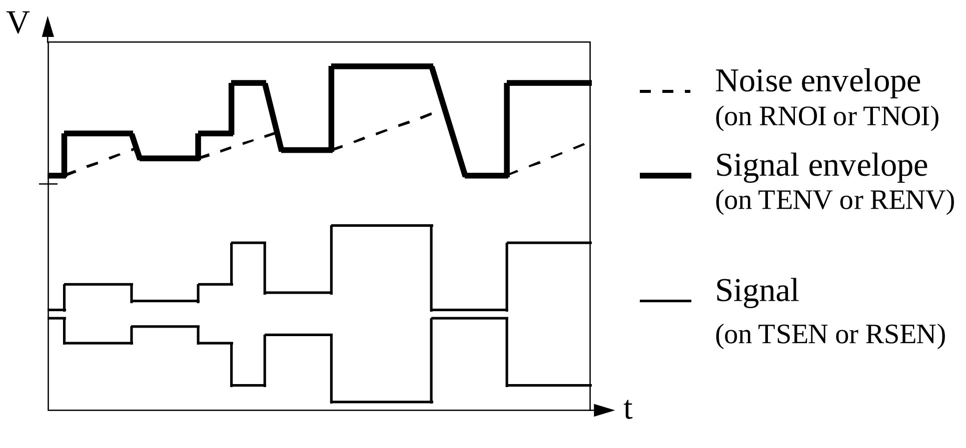


Fig. 3.4.2 Typical behavior of the signal and noise detectors

In fig.3.4.2 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.

### 3.4.2 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 TEA1095 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.3.4.1, 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 13mV. This so called speech/noise threshold is implemented in both the receive and the transmit channel. At the end of paragraph 3.4.4 a way to increase this threshold is discussed.

Second, the signal on TXIN contains both 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 TEA1095 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 TEA1095 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.3.4.1, is adjustable by Rsen.

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.3.4.3. 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.3.4.3 Truth table of the detection logic

When pin MUTETX is made high, see paragraph 3.2.2, the output current is forced to be +10 µA, which forces the TEA1095 into Rx-mode and mutes the microphone amplifier. When pin MUTERX is made high, see paragraph 3.3.3, the output current is forced to be -10 µA, which forces the TEA1095 into Tx-mode and mutes the receive path. When both MUTETX and MUTERX are made high, both channels are muted.

The voltage on pin SWT is internally limited to IDT-0.4 V and IDT+0.4 V.

### 3.4.3 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 will be described with the help of fig.3.4.4.

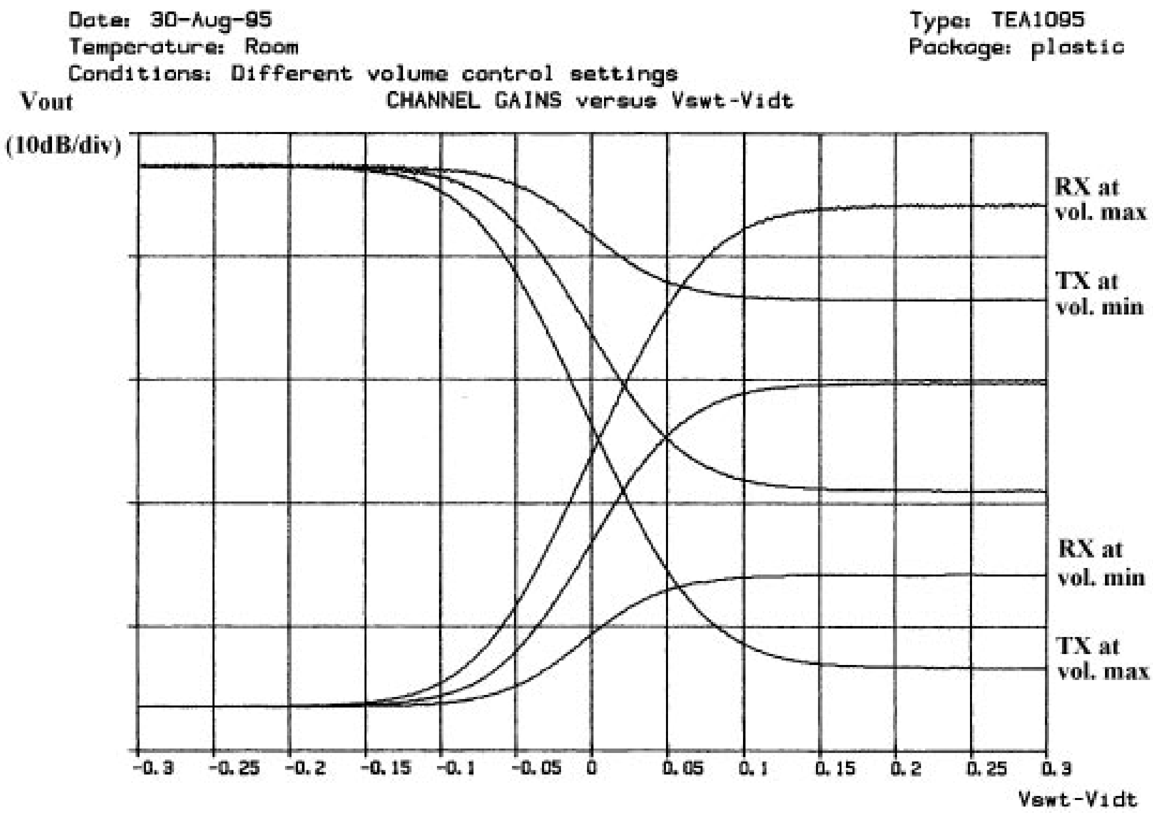


Fig. 3.4.4 Behavior of the voice switch

When the voltage on SWT is more than 180 mV below the voltage on IDT, the TEA1095 is fully switch 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 TEA1095 is fully switch to Rx-mode (gain of the receive path at maximum and gain of the transmit path at minimum). The TEA1095 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 3.4.4. Both Rswr and Rstab set internally used reference currents which are proportional to absolute temperature (PTAT).

As already stated in &3.3 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 TEA1095 virtually does not switch over. In order to avoid inversion of the gain in Rx-mode, the volume control range of the TEA1095 cannot be larger than the switching range.

### 3.4.4 Adjustments and performances

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

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 taken 3.65 kΩ. The value of the resistor Rswr can vary between 3.65 kΩ and 1.5 MΩ resulting in a switching range between 0 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 = Atx1095 + Atx1096 + Ast + Arx1096 + Arx1095 + Als1096 + Aac - Asw.

with Atx1095 = sending gain of the TEA1095 (TXIN to TXOUT)

Atx1096 = sending gain of the TEA1096 (MIC to LN)

Ast = electrical sidetone

Arx1096 = receive gain of the TEA1096 (LN to QR)

Arx1095 = receive path gain of the TEA1095 (RXIN to RXOUT)

Als1096 = loudspeaker amplifier gain of the TEA1096 (LSI to QLS)

Aac = electro-acoustic coupling from loudspeaker to microphone (QLS 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 10 to 15 dB ).

The electrical sidetone is the difference (in dB) between the wanted receive signal on the TEA1096 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 lenght, 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.

In appendix A, a method for measuring the required switching range, based on the above calculation, is given.

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

As for the calculation, it is necessary to identify what are the worst conditions 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 10 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 is determined by the value of Rrsen according to:

Vdialtone = 4.2 µA * Rrsen

With Rrsen of 10 kΩ, the dial tone detector level will be 42 mVrms. This means, a continuous signal on the input RXIN larger than 42 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 choosen 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 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) - Atx1095 - Atx1096 - Ast - Arx1096 + Atsen + 1/2 Aloop

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

In this relation, the maximum loop gain and the worst case sidetone are used. If it is prefered 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).

In appendix A, a method for determining the required Rtsen, based on the above calculation, is given.

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 $\cong$ 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 (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. Suppose the set is in full Tx-mode, then the voltage on SWT will be V(IDT)-400 mV, see fig.3.4.4. 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

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 TEA1095 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.3.4.5 can be applied.

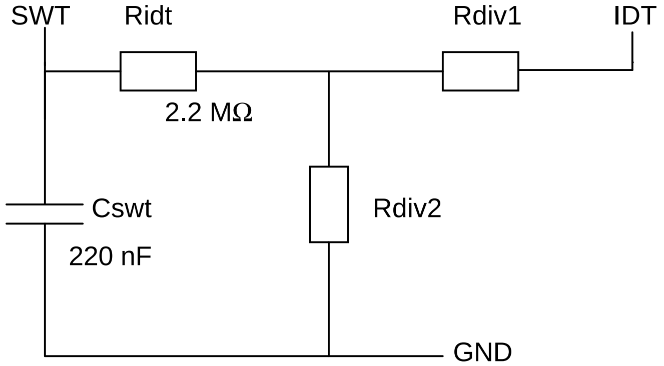


Fig.3.4.5 Circuit for shifting the idle mode

With the circuit of fig.3.4.5, 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.3.4.4 , 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 inorder to limit the current drawn from IDT. By connecting Rdiv2 to VBB instead of GND, the idle mode is shifted towards the receive mode.

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 TEA1095, 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 serie 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 depicted in fig.3.4.1, 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 15 kΩ.

## 4. APPLICATION COOKBOOK

In this chapter, the procedure for making a basic application with a speech and listening-in IC TEA1096 and the handsfree duplex controller TEA1095 will be given. With the help of fig. C1 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. The preferred values are given between brackets.

The application of fig. C1 is a basic application of the TEA1095 voice switched speakerphone IC with the TEA1096 speech and listening-in circuit.

As can be seen in fig. C1, only a few components have a fixed value. All other values will follow from the cookbook of fig.4.1.

| Step | Adjustment |
|---|---|
| **DC setting** : | |
| Adjust the DC setting of the TEA1096 to the local PTT requirements. | |
| Voltage LN-VEE | |
| DC slope | Refer to [4] and/or to [12] (N° : ETT94001) |
| Supply point VCC | |
| Supply point VBB | |
| Artificial inductor | |
| **Impedance and sidetone** : | |
| After setting the required set impedance, the sidetone has to be optimized using the two sidetone netwoks in order to minimize the loop gain in all line conditions. AGC can be adjusted at that step. | |
| Set active impedance | Refer to [4] and/or to [12] (N° : ETT94001) |
| Sidetone | |
| AGC | |
| **TEA1096 transmit and receive gains** | |
| Transmit gain | Total transmit and receive gains have to be siplitted between TEA1096 and TEA1095. It is suggested to leave 15dB of transmit gain for the TEA1095 and to adjust the receive gain between LN and QRP at -3 dB.<br>Ctxo and MICP input impedance form a high-pass filter.<br>A capacitor in parallel with the transmit gain resistor (between TEA1096 pins 17 and 11) form a low-pass filter.<br>A resistor bridge can be inserted between TXOUT of TEA1095 and MICP of TEA1096. |
| Receive gain | A capacitor in parallel with the receive gain resistor (between TEA1096 pins 26 and 25) form a low-pass filter.<br>For more details, refer to [4] and/or to [12] (N° : ETT94001) |
| **TEA1095 microphone amplifier** (see & 3.2): | |
| After the sensitivity and the curve of the microphone are adjusted, the gain can be adjusted to the desired value | |
| Microphone sensitivity | Rmic sets the sensitivity. Together with Rmf, Rmic provides the polarisation of the electret. |
| Frequency curve | Cmic with Rmic and the output impedance of the electret form a low-pass filter.<br>Ctxin with the 20 kΩ input impedance at TXIN form a high-pass filter. |
| Transmit gain and stability | Rgatx sets the microphone amplifier gain : Atx=20*log(0.72*Rgatx/Rstab)<br>Cgatx may be necessary for stability and forms a low-pass filter with Rgatx. |

| **Step** | **Adjustment** |
|---|---|
| **TEA1096 loudspeaker amplifier** : | |
| The gain is fixed at 35.5 dB , a high-pass filter can be made, the dynamic limiter timing can be chosen. | |
| Frequency curve | Crxo and/or capacitor in series with loudspeaker can form high-pass filters |
| Dynamic limiter timing | Capacitor at pin DLL. |
| **TEA1095 receive channel** (see & 3.3) : | |
| The gain of the receive pass and the curve can be adjusted. The volume control range can be chosen. | |
| Receive gain | Rgarx : the total receive gain of the set is equal to receive gain of the TEA1096 (-3 dB ?), the gain of the loudspeaker amplifier (35.5 dB) and the gain set with Rgarx. |
| Receive curve | Crxi with the input impedance of 20 kΩ at pin RXI form a high-pass filter (a cut-off frequency between 100 and 200 Hz is advised), |
| Volume control | Cgarx in parallel with Rgarx forms a low-pass filter.<br>A linear potentiometer of 10 kΩ is suggested (3 dB for each 950 Ω). |
| **TEA1095 Duplex controller** (see & 3.4) : | |
| 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=Atx1095+Atx1096+Ast+Arx1096+Arx1095+Als1096+Aac-Asw < 0dB<br>Choose Asw with safety margin<br>Adjust Rswr : Asw=20log(Rswr/Rstab) with Rstab fixed at 3.65kΩ |
| Dial tone detector | Rrsen : Vdialtone=4.2 µA*Rsen |
| Sensitivities | Rtsen for balanced sensitivities between Tx and Rx<br>Ctsen form a high-pass filter with Rtsen<br>Crsen form a high-pass filter with Rrsen |
| Signal envelopes | Ctenv (0.47 µF), Crenv (0.47 µF) :<br>maximum attack : 120 µ/(3*Cenv) (dB/ms), release : 1 µ/(3*Cenv) (dB/ms) |
| Noise envelopes | Ctnoi (4.7 µF), Crnoi (4.7 µF) :<br>attack : 1 µ/(3*Ctnoi) (dB/ms), maximum release : 120 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. 4.1 Steps in the design flow of the TEA1095 with TEA1096

## 5. APPLICATION EXAMPLE

In this chapter, a basic application example of the TEA1095 with the TEA1096 is depicted. Only the essential elements are given, for instance no bridge, no protection, no ringer, no line interruptor are included.

Fig. C2 gives the basic handsfree application of the TEA1095 together with the speech and listening-in circuit TEA1096. This application doesn't include handset telephony (which would have to be added by means of analog switches) but only basic handsfree telephony.

- **DC setting** : nominal DC characteristic and nominal $V_{BB}$ of 3.6 V are chosen.

**- Impedance and sidetone** : the impedance is a complex one (220 Ω, 825 Ω//115 nF). A standard 600 Ω impedance would save six components.

**- TEA1096 transmit and receive gains** : the transmit gain of TEA1096 is set at 30.5dB by means of a resistor bridge (22 kΩ, 7.5 kΩ at microphone input combined with an internal gain of 43 dB set by 32.5 kΩ). The cut-off frequency of the high-pass filter is set at 110 Hz by 2*100 nF in series, the cut-off frequency of the low-pass filter is set at 7.2 kHz by 680 pF//32.5 kΩ.

The receive gain of TEA1096 is set at -3 dB by 82.5 kΩ. The cut-off frequency of the low-pass filter is set at 4.1 kHz by 470 pF//82.5 kΩ.

-**TEA1095 microphone amplifier** : the sensitivity of the electret microphone is set by Rmic=1.5 kΩ. The microphone power supply is filtered by Rmf, Cmf with a cut-off frequency of 16Hz. An electrical low-pass filter is provided with Cmic, in combination with the effect of the cabinet and the self curve of the electret, it adjusts the high frequency part of the transmit curve. A high-pass filter is formed with Ctxin and the 20 kΩ input impedance at TXIN, its cut-off frequency is set at 530 Hz (a higher cut-off frequency may be necessary to better minimize the room echo and/or to better compensate the line attenuation).

The gain of the microphone channel is set at 15.5 dB by Rgatx=30.1 kΩ, a low-pass filter with Cgatx=2.2 nF would provide a cut-off frequency of 2.4 kHz.

**- Loudspeaker amplifier** : nominal loudspeaker amplifier application is made (gain of 35.5 dB).

**- TEA1095 receive channel** : the gain of the receive path of the TEA1095 is set at -7 dB by Rgarx=3.65 kΩ ( the total receive gain of the set is : 35.5 - 3 - 7 = 25.5 dB). A high-pass filter with a cut-off frequency of 120 Hz is formed by Crxin=68nF and the input impedance of 20 kΩ at RXIN. A low-pass filter with Cgarx=15 nF would provide a cut-off frequency of 2.9 kHz for the adjustment of the loudspeaker curve.

A 10kΩ linear potentiometer provides a volume control range of about 31dB.

**- Duplex controller** : after optimized acoustic adaptation (**reduction of acoustic coupling between loudspeaker and microphone** and good sidetone adaptation) the switching range is set at 40dB by Rswr=365k, including safety margin.

With Rrsen=10 kΩ, the dial tone detector level is set at 42 mV which means 60 mVrms on line.

With Crsen=100 nF, the cut-off frequency of the high-pass filter is 160 Hz.

Rtsen and Ctsen respectively equal Rrsen and Crsen in order to balance transmit and receive sensitivities.

With Ctenv and Crenv=0.47 µF, the max. attack time of the signal envelopes is set at 85 dB/ms.

With Ctnoi and Crnoi=4.7 µF, the max. attack time of the noise envelopes is set at 0.07 dB/ms.

The switch-over timing is nominally set by Cswt=220 nF and the idling timing by Ridt=2.2 MΩ.

Due to its large flexibility ( range of adjustments and architecture ), the TEA1095 can be implemented in a lot of environments, offering handsfree function with various speech interfaces and loudspeaker amplifiers, with or without external supply.

## 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 and Rgarx has to be done with very short traces (specially STAB input which sets all the gains must be very immune).

VOL and TXI inputs may also be sensitive (RF signals entering these pins would be amplified). Rvol can be connected with short traces if possible or VOL input may be lightly decoupled by a capacitor to GND. Care has to be taken with the lay-out of the microphone amplifier, which is also helpfull for the noise, providing a good decoupling to GNDTX. A low-pass RC filter may be added at the input of the amplifier.

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

RXIN input may also be decoupled to GND.

A low impedance capacitor in parallel with the electrolythic one between VBB and GND may help.

## 7. REFERENCES

[1] TEA1095 Voice Switched Speakerphone IC

Device specification

[2] Philips Semiconductors

SEMICONDUCTORS FOR TELECOM SYSTEMS - IC03 -

[3] TEA1094 Handsfree IC

Device specification

[4] TEA1096 / TEA1096A Line interface and Listening-in ICs

Device specification

[5] TEA1081 Supply circuit with power-down for telephone set peripherals

Device specification

[6] TEA1083 / TEA1083A Call progress monitor for line-powered telephone set

Device specification

[7] TEA1085 / TEA1085A Listening-in circuit for line-powered telephone set

Device specification

[8] TEA1112 / TEA1112A / TEA1113 Low voltage versatile telephone transmission circuits with dialler interface

Device specifications

[9] TEA1062 / TEA1062A Low voltage versatile telephone transmission circuits with dialler interface

Device specification

[10] TEA1064A / TEA1064B Low voltage versatile telephone transmission circuits with dialler interface and level dynamic limiting

Device specification

[11] Application of the TEA1094 handsfree circuit ( ETT/AN94004 )

[12] Application of the TEA1096 transmission and listening-in circuit ( ETT94001 )

[13] Measures to meet EMC requirements for TEA1060-family speech transmission circuits ( ETT89016 )

[14] Application of the speech transmission circuit TEA1062 ( ETT89008 )

[15] Application of the versatile speech/transmission circuit TEA1064 in full electronic telephone sets ( ETT89009 )

[16] Philips Semiconductors
Wirebound telecom
APPLICATIONS HANDBOOK

## APPENDIX A MEASUREMENT SETUP

This appendix describes how the loop gain and the sensitivity setting can be measured. These measurements provide a base for obtaining of the final values which, however, can only be reached by performing subjective tests.

For determining the loop gain, the worst case for both acoustical coupling and sidetone must be considered. This can be done with the measurement setup of fig. A.1.

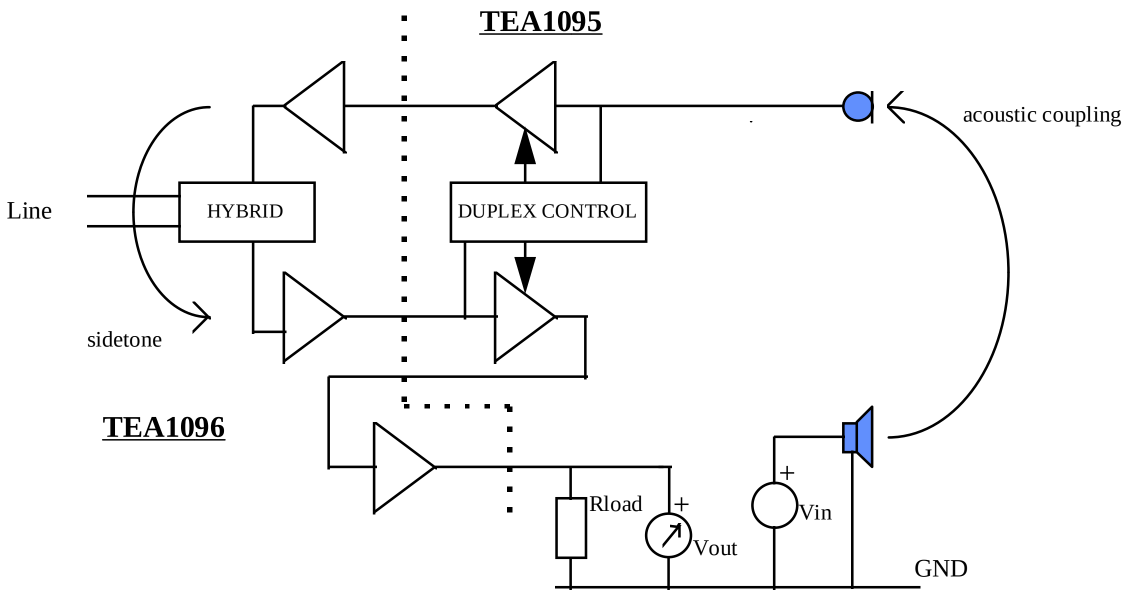


Fig. A. 1 Setup for loop gain measurement

The loudspeaker is disconnected from the set and an electrical signal is applied to it. The acoustical signal coming from the loudspeaker is coupled to the microphone (Aac), transmitted to the line (Atx1095 + Atx1096), returns via sidetone (Ast) and is amplified again to the loudspeaker output of the TEA1096 (Arx1096 + Arx1095 + Als1096). The output is loaded with a loudspeaker equivalent impedance (Rload), it has to be checked that the TEA1096 loudspeaker amplifier is not saturated and that its gain is not reduced by the dynamic limiter. The total gain of this loop is reduced with the switching range (Asw).

The gain from the loudspeaker connection (Vin) to the loudspeaker amplifier output (Vout) is the loop gain (Aloop), given as:

20 * log ( Vout / Vin ) = Aloop

Aloop = Aac + Atx1095 + Atx1096 + Ast + Arx1096 + Arx1095 + Als1096 - Asw

This measurement has to be carried-out with a Vin level avoiding saturation of the loudspeaker amplifier (start with 100 mVrms) in the frequency range 200 Hz to 5000 Hz (100 to10000 Hz for high quality aparatus).

From the curve obtained, the maximum gain (<0 dB) indicates the gain margin.

Sidetone curve changes with line impedance, so, the Aloop curve will change in the same way. It is advisable to double-check with different line conditions (impedance and length) that the identified worst line condition is really the worst all over the frequency range.

Moving around the set also modifies acoustic coupling, so, it is advised to proceed as for sidetone.

The measurement combining both worst case conditions of sidetone and acoustical coupling gives the worst case loop gain and thus the gain margin.

All transmit and receive gains and curves are supposed to be set, the value of the switching range however, can only be finalized after the measurement. Therefore, during the measurement, the switching range must be set to a preliminary value. An advised preliminary value for measurement is 40 dB.

When the correct switching range is set, the sensitivity settings can be determined.

Rrsen is determined to get the correct tone detector level.

For determining the sensitivity, calculations can be done as described in &3.4.4. It is also possible measuring the correct sensitivity setting with measurement setup of fig. A. 2.

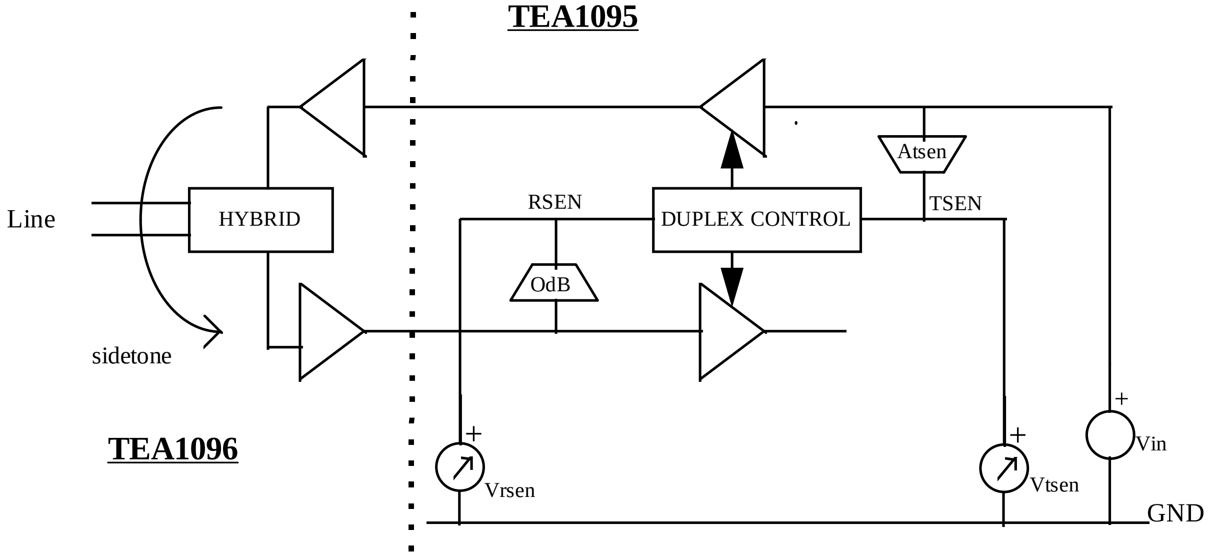


Fig. A. 2 Simplified measurement setup for determining sensitivity

Both the loudspeaker and microphone are disconnected from the set. An input signal (Vin) of 1 mVrms is applied to the microphone input. This signal is amplified to pin TSEN (Atsen), it is also transmitted to the line, returns via sidetone, is amplified by the TEA1096 to the TEA1095 pin RXIN and buffered to pin RSEN. The ratio of the signal on pin TSEN (Vtsen) and RSEN (Vrsen) is given as :

20 * log (Vtsen/Vrsen) = Atsen - Atx1095 - Atx1096 - Ast - Arx1096

This measurement has to be done in nominal line and environment conditions in the frequency range 300 Hz to 3400 Hz. The TEA1095 must be forced in Tx-mode by making pin MUTERX high, the receive channel is muted but Vrsen is not affected.

Out of the measured ratio of Vtsen and Vrsen it follows :

20 * log (Rtsen / Rrsen) = 20 * log (Vtsen / Vrsen) + Aloop / 2 ( see &3.4.4, Ad.3)

Both the ratio of Vtsen and Vrsen and Aloop are frequency dependent, so the optimal value of Rtsen would be frequency dependent. However, this can be simplified by using the lowest value of 20*log(Vtsen/Vrsen) and the highest value of Aloop to obtain the correct Rtsen.

The value of Rtsen found is a starting point for obtaining the final value which can only be reached by subjective tests with real telephone conditions.

## APPENDIX B LIST OF ABBREVIATIONS AND DEFINITIONS

| | |
|---|---|
| Aac | Electro-acoustic coupling ( electrically measured ) |
| AGC | Automatic line loss compensation of the TEA1096 |
| Aloop | Loop gain of a handsfree telephone set |
| Als1096 | TEA1096 loudspeaker amplifier gain |
| Arx1095 | Gain of the receive path of TEA1095 |
| Arx1096 | Receive gain of TEA1096 |
| Ast | Sidetone gain |
| Asw | Switching range |
| Atsen | Gain from TXIN to TSEN of 40dB |
| Atx1095 | Gain of the transmit path of TEA1095 |
| Atx1096 | Transmit gain of the TEA1096 |
| Cgarx | Capacitor setting receive path amplifier low-pass filter |
| Cgatx | Capacitor setting microphone amplifier low-pass filter |
| Cmf | Microphone supply filter capacitor |
| Cmic | Microphone low-pass filter capacitor |
| Crenv | Capacitor determining the receive signal envelope |
| Crnoi | Capacitor determining the receive noise envelope |
| Crsen | DC blocking capacitor of receive sensitivity setting |
| Crxi | Receive input capacitor |
| Crxo | Receive output capacitor |
| 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 | Microphone amplifier input capacitor |
| Ctxo | Transmit output capacitor |
| dBmp | dBm psophometrically weighted (0dBmp=1mW) |
| $\delta$Vswt | Voltage difference on SWT |
| GARX | Receive gain adjustment pin |
| GATX | Transmit gain adjustment pin |
| GND | Ground reference pin |

| | |
|---|---|
| GNDTX | Ground reference pin for microphone signals |
| IDT | Idle-mode timing adjustment pin |
| Iswt | Output current through pin SWT (from logic) |
| Ix-mode | Idle mode |
| LN | Positive line terminal of TEA1096 or TEA106x |
| LSI | Loudspeaker amplifier input of TEA1096 |
| MICP,MICM | Microphone input of TEA1096 |
| MOSFET | Meta Oxide Field Effect Transistor |
| MUTERX | Receive channel mute |
| MUTETX | Transmit channel mute |
| NC | not connected |
| PD | Power-down input pin (reduced power consumption) |
| PTAT | Proportional to absolute temperature |
| PTT | Public telephone company |
| QRP | Earphone amplifier output of TEA1096 |
| RENV | Receive signal envelope timing adjustment pin |
| RFI | Radio frequency interference |
| Rgarx | Resistor setting receive path amplifier gain |
| Rgatx | Resistor setting transmit path amplifier gain |
| Ridt | Resistor setting Ix-mode timing |
| Rload | Loudspeaker equivalent load resistor |
| Rmf | Microphone supply filter resistor |
| Rmic | Resistor setting microphone sensitivity |
| RNOI | Receive noise envelope timing adjustment pin |
| 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 |
| Rvol | Volume control potentiometer |
| RXIN | Receive path input |
| RXOUT | Receive path output |

| | |
|---|---|
| Rx-mode | Receive mode |
| SLPE | DC slope pin of TEA1096 or TEA106x |
| STAB | Reference current pin |
| SWR | Switching range adjustment pin |
| SWT | Switch-over timing adjustment pin |
| TENV | Transmit signal envelope timing adjustment pin |
| Tidt | Idle mode timing |
| TNOI | Transmit noise envelope timing adjustment pin |
| TSEN | Transmit signal envelope sensitivity adjustment pin |
| TXIN | Microphone amplifier input |
| TXOUT | Transmit path output |
| Tx-mode | Transmit mode |
| $V_{BB}$ | Positive supply input of TEA1095 |
| Vdt, Vdialtone | Dial tone detector level |
| VEE | Reference pin on TEA1096 |
| VOL | Volume adjustment pin |

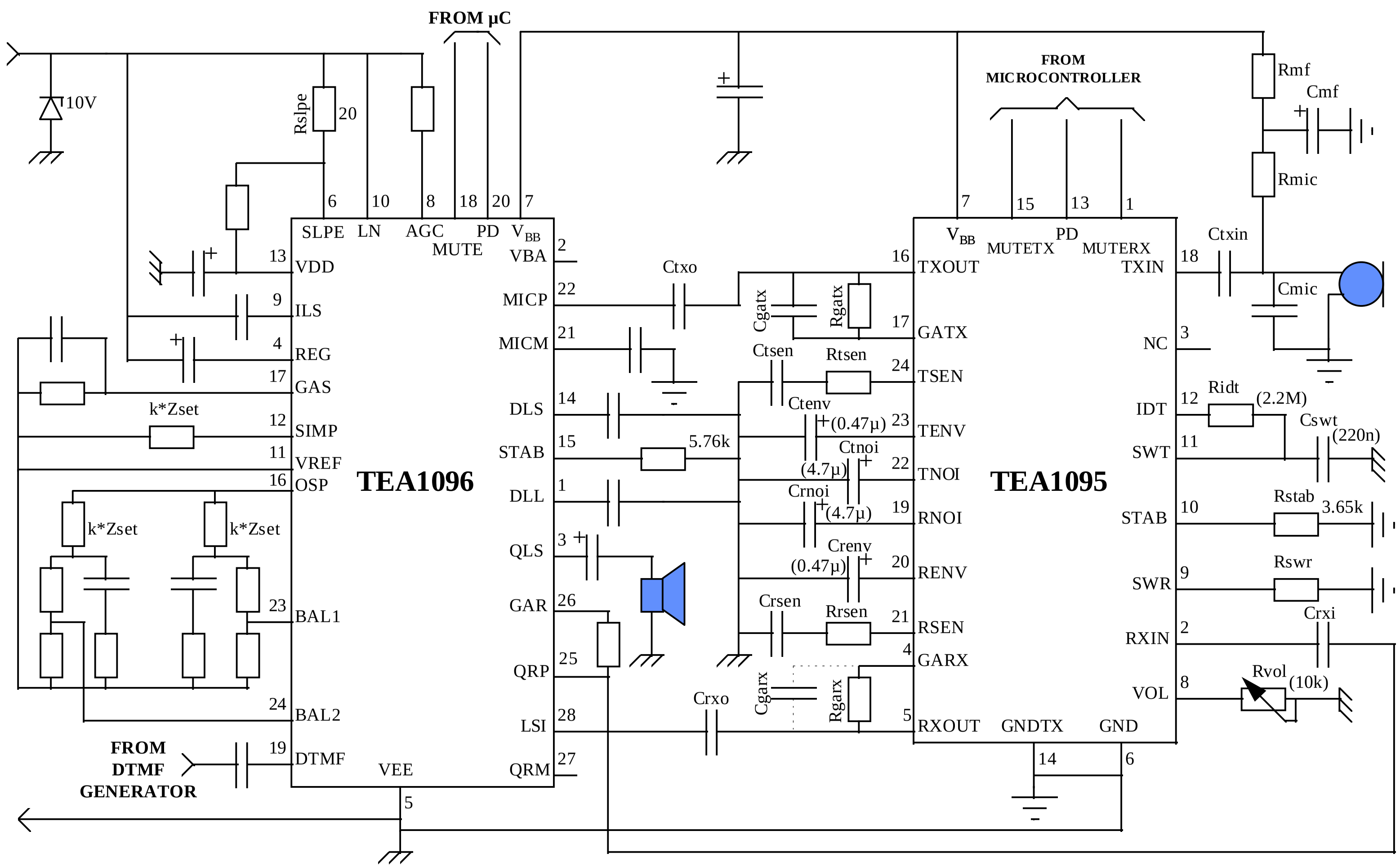


Fig. C1 Basic application of TEA1095 with TEA1096

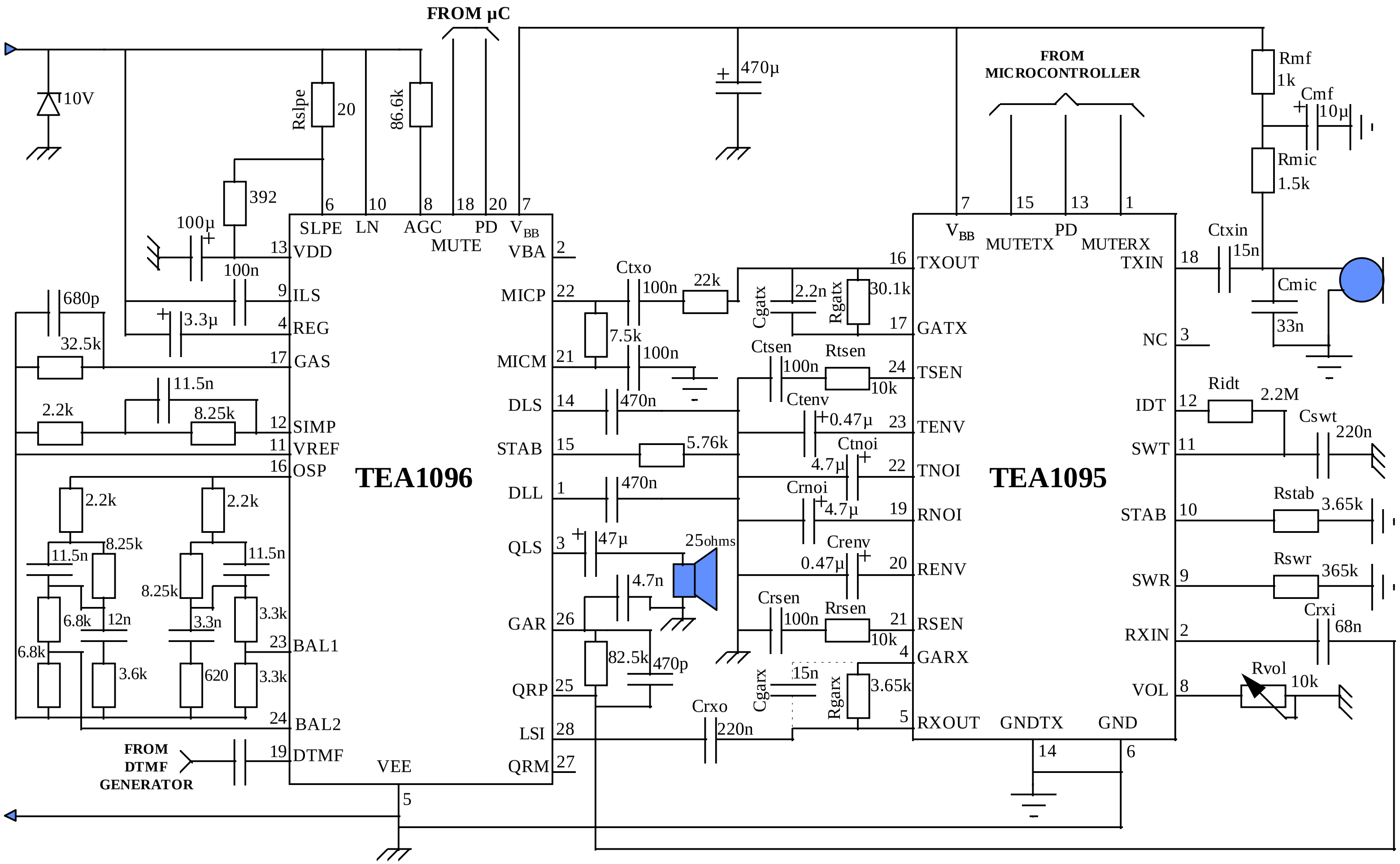


Fig. C2 Basic application of TEA1095 with TEA1096