SGS-THOMSON MICROELECTRONICS

APPLICATION NOTE

TEA7540 - SPEAKERPHONE CIRCUIT FOR HANDSFREE TELEPHONE SET

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I - INTRODUCTION

This note discusses the principles and feature of a hands-free telephone set using TEA7540, the description of TEA7540 IC, design applications, and the set-up procedure.

Supply : TEA7540 can be :

- powered directly from the telephone line due to its low consumption (2.1mA typically) and minimum operating voltage (2.5V).
- powered from a stabilized voltage source from 2.5V to 7V. The current consumption is from 2.1mA up to 2.8mA when the voltage supply varies from 2.5V to 7V.

Duplex voice paths :

The handset allows 2-way simultaneous conversations (transmit and receive). This is commonly called the ”full duplex operation”.

In the hands-free telephone set case, the loop that exists within the system forces the introduction of an attenuation into the path which is detected as the dominated voice path. Only one person speaks at a time, while the other listens. This is the case of current speakerphones available on the market. They are called ”half duplex operating systems”. TEA7540 performs the switching function needed for speakerphone operation. It allows a good approximation of full duplex operation by attenuation switching optimization and by gain control for the transmit and receive voice paths.

TEA7540 has also a system which enables constant background noise to be ignored in making the transmit/receive switching decision.

TEA7540 voice-switched speakerphone circuit incorporates compressor-amplifiers, attenuators, level detectors, and a switching control system. It can also be operated in the handset mode with a chip select command which bypasses the switching functions associated with the hands-free mode.

Each input of TEA7540 includes a compressor pre-amplifier. This feature allows the transmitted line level to be independent of the distance between the speaker and the microphone. Similary, the received level does not depend on the telephone line charcateristics.

Specially designed for basic low cost hands-free telephone set applications, TEA7540 provides a control interface between the speech IC, the hands-free microphone, and the loudspeaker IC. The interface can also be extended to a microcomputer with more sophisticated operating facilities and controls.

II - ARCHITECTURE DESCRIPTION

II.1 - Reference Schematics

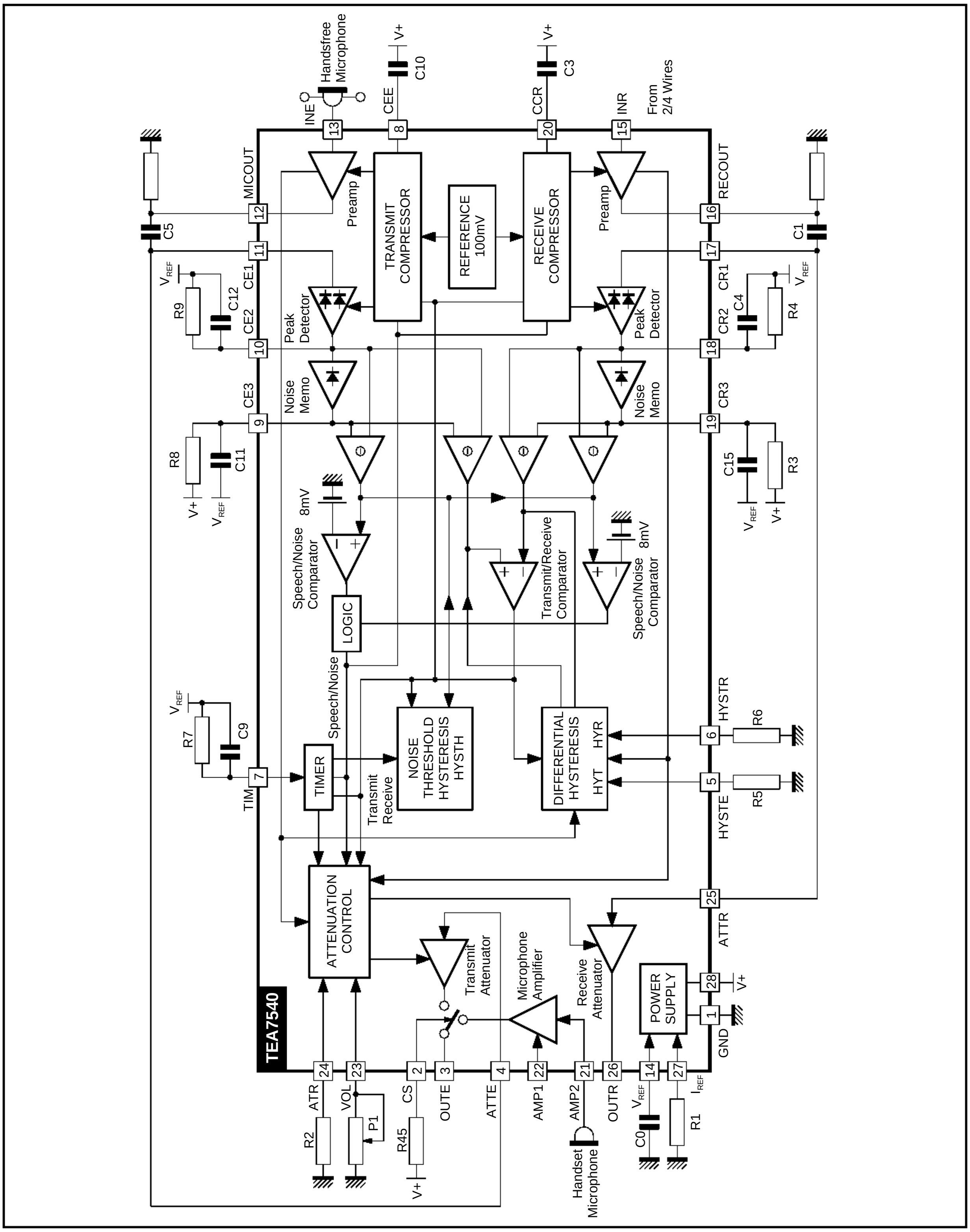
II.1.1 - TEA7540 Pin Description

No	Pin	Description
1	GND	Ground
2	CS	Chip Select
3	OUTE	Transmit Attenuator Output
4	ATTE	Transmit Attenuator Input
5	HYSTE	Transmit Hysteresis
6	HYSTR	Receive Hysteresis
7	TIM	Timer
8	CCE	Transmit Compressor Time Constant
9	CE3	Transmit Background Noise Memorization
10	CE2	Transmit Peak Detector
11	CE1	Transmit Rectifier Input
12	MICOUT	Transmit Compressor Output
13	INE	Transmit Compressor Input
14	VREF	Reference Voltage
15	INR	Receive Compressor Input
16	RECOUT	Receive Compressor Output
17	CR1	Receive Rectifier Input
18	CR2	Receive Peak Detector
19	CR3	Receive Background Noise Memorization
20	CCR	Receive Compressor Time Constant
21	AMP2	Microphone Pre-amplifier Input
22	AMP1	Microphone Pre-amplifier Supply
23	VOL	Volume Control
24	ATR	Attenuator Control
25	ATTR	Receive Attenuator Input
26	OUTR	Receive Attenuator Output
27	IREF	Reference Current
28	V+	Power Supply Input

II - ARCHITECTURE DESCRIPTION (continued)

II.1.2 - TEA7540 Block Diagram

Figure 1



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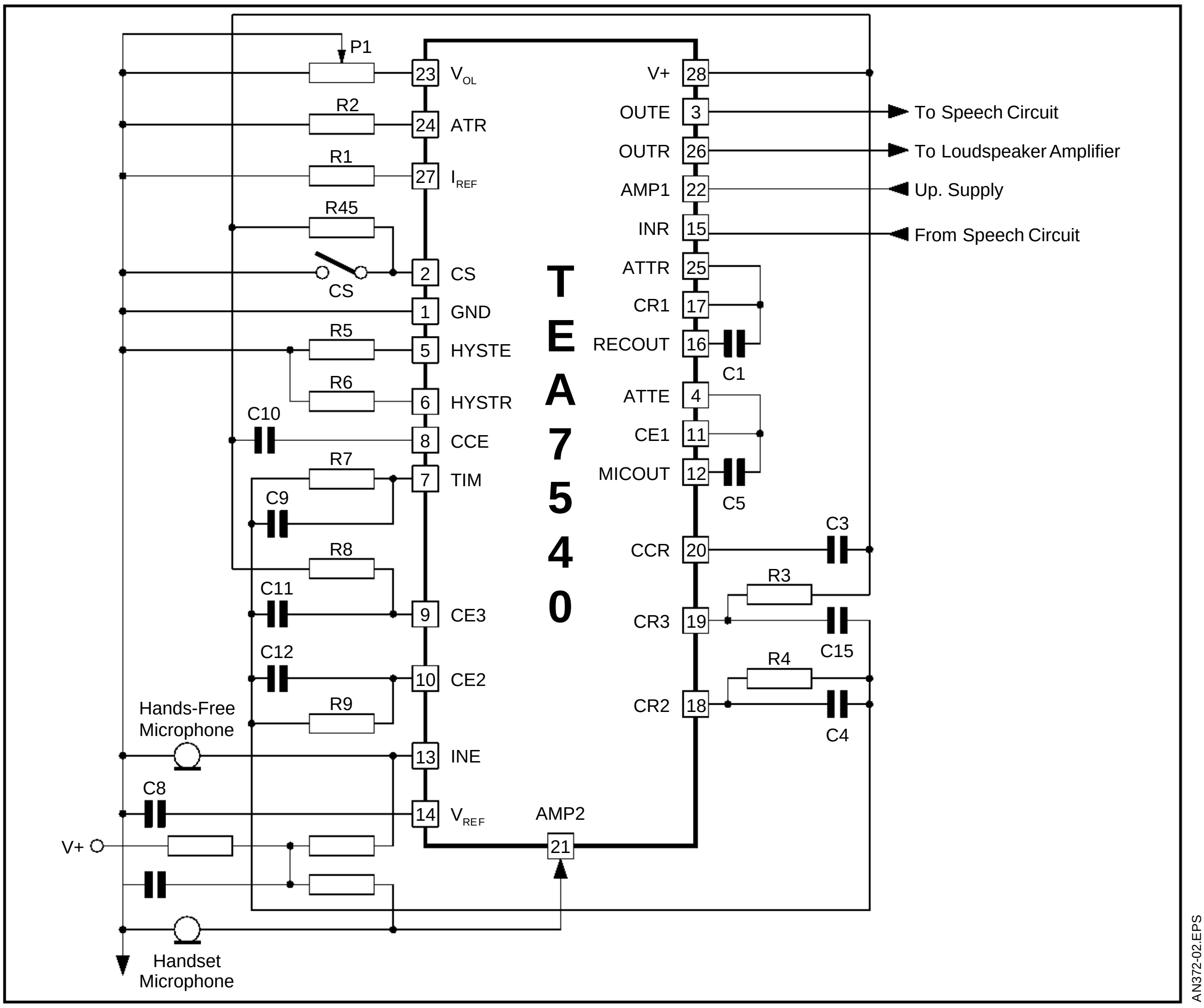
II - ARCHITECTURE DESCRIPTION (continued)

II.1.3 - TEA7540 Typical Application

For more information, see Application Section.

Figure 2



II.2 - Main Characteristics of a Telephone Set using TEA7540

TEA7540 is easily supplied from the line.

TEA7540 has an automatic level control, called compressor, in each channel : TEA7540 is self-adaptive depending on line (receive line losses due to line length variations) and room conditions (distance variations between microphone and speaker).

TEA7540 is not affected by typical noise conditions in each path.

Programmable attenuators are provided separately in each voice path.

Chip select allows hands-free and monitoring operation separately.

Microphone preamplifier for handset mode is provided on-chip.

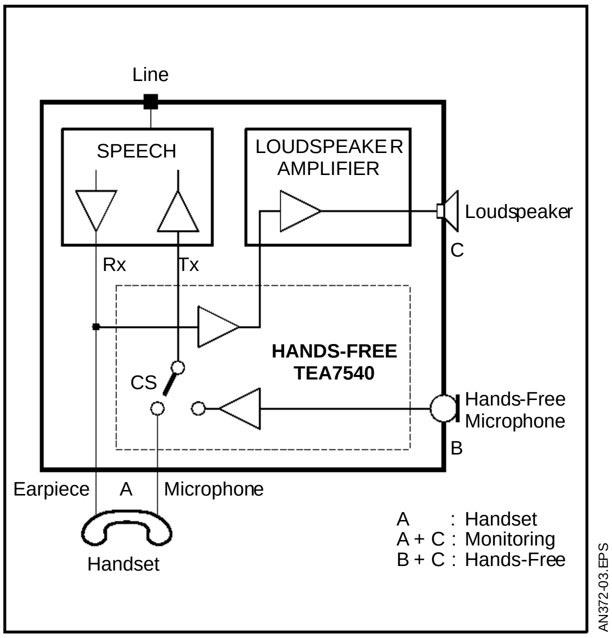
Different national standards are easily matched by varying the values of a small number of external components.

II - ARCHITECTURE DESCRIPTION (continued)

II.3 - Hands-free Telephone Set Architecture

A complete hands-free telephone set design incorporates the following circuits :
- TEA7540
- speech circuit
- loudspeaker amplifier

Figure 3 : Simplified Schematic of an Hands-free telephone set



The chip select switch (Pin2) is used to select one of the two basic operating modes of TEA7540 :

Handset Mode

a) For monitoring operation in the receive path is introduced a programmable gain and in the transmit one a fixed gain (20dB).
b) The handset microphone is enabled and the hands-free microphone is disabled.

Hands-Free Mode

a) The handset microphone is off and the handsfree microphone is on.
b) The attenuation controls are operative in the transmit and receive paths.

II.4 - Schematic Blocks and Operative Modes

The circuit basically contains 5 blocks : (Figure 4)
- Tx compressor : transmit pre-amplifier with variable gain
- Tx attenuator: programmable transmit attenuator
- Rx compressor : receive pre-amplifier with variable gain
- Rx attenuator: programmable receive attenuator
- Control block : attenuation control system

The control block monitors the level of transmit/receive signals and provides a set of control signals which switch the system into one of the three following operational modes :

Transmit Mode

- The Tx attenuator has a constant attenuation (0dB),
- The Tx compressor maintains constant its output V1 (200mV_{PP}),
- The Rx attenuation provides the programmed attenuation,
- The Rx compressor is disabled.

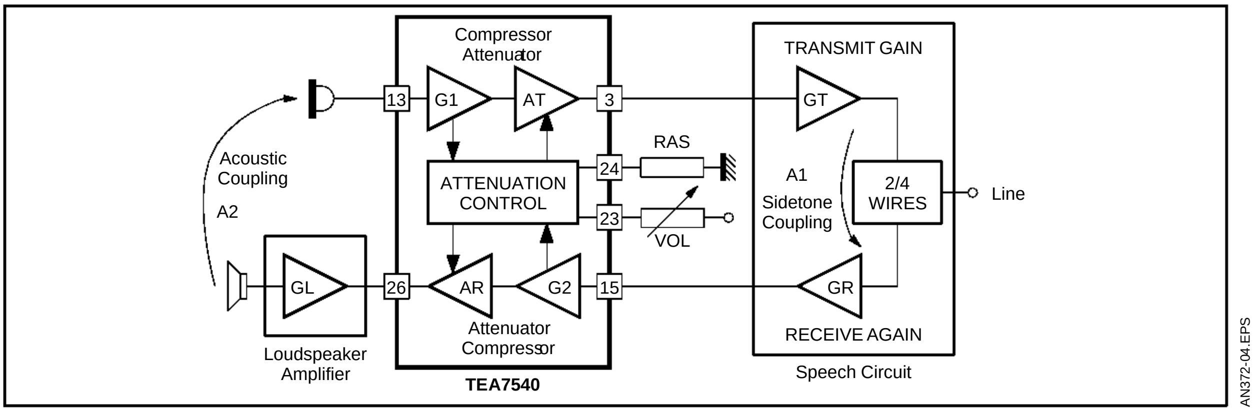
Receive Mode

- The Rx attenuator has a constant attenuation (0dB),
- The Rx compressor maintains constant V2 (200mV_{PP}),
- The Tx attenuation provides the programmed attenuation,
- The Tx compressor is disabled.

Idle Mode

- The loop attenuation (AS, programmable by Pin 24) is distributed equally (AS/2) between the transmit and receive attenuators.

Figure 4 : Loop Gain Principle



II - ARCHITECTURE DESCRIPTION (continued)

II.5 - Additional Features of TEA7540

II.5.1 - Low Voltage Operation and Power Saving with TEA7540

a) Efficient operation over a wide range of supply voltage (2.5V to 7V) to fulfil different national standards. The consumption is at low current even at maximum supply voltage ensuring low power consumption. In some countries the voltage available over the telephone set is as low as 5V and the minimum line current is lower than 20mA. TEA7540 allows line powered application due to its minimum operating voltage, 2.5V, and operating current, 2.1mA.

b) Background noise immunity :
TEA7540 has a system which enables constant background noise to be ignored in making the Selection of the appropriate operating mode : transmit/ receive/ idle (see II.5.4 section). The background noise control of TEA7540 keeps the system in idle mode until the level of the signal is higher than the noise threshold, in this case the system switches in receive or transmit mode.

c) Half duplex/full duplex :
TEA7540 allows a good approximation of full duplex operation by attenuation switching optimization and by gain control for the transmit and receive paths.

d) Stability :
The attenuation control of TEA7540 distributes the required loop attenuation between the transmit or receive channels according to the mode of operation: transmit receive or idle. Attenuation is required to avoid instability which appears when the total loop gain (electronic gain + transducer + acoustic coupling gains) is higher than 0dB.

Stability condition : (GR+GT)<<(AA+AE)
(see Figure 5)

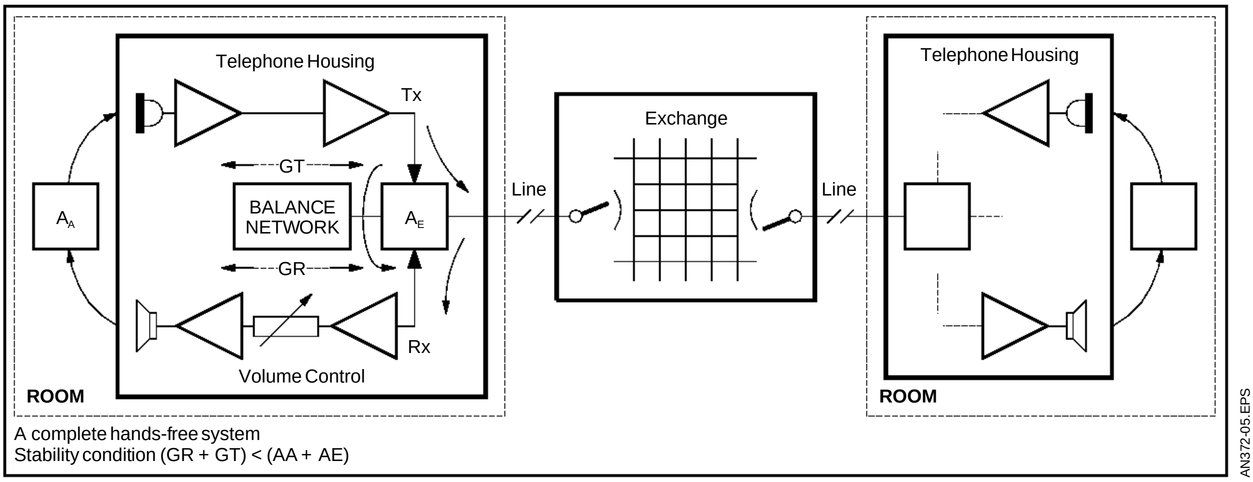
II.5.2 - Automatic Level Control with TEA7540

TEA 7540 has two signals compressors which perform an automatic level control of the Tx and Rx signals and provide constant amplitude voltage outputs (200mV peak-peak).

a) The transmit compressor compresses the hands-free microphone signal in such a way that the distance between the speaker and the microphone can be from 0.5 up to 3 meters. This implies : see Figure 6.

b) The line level speech depends on the line length. The receive compressor compresses the receive signal from speech circuit to compensate the looses due to line length. This implies : see Figure 7.

Figure 5 : Howling effect due to acoustic coupling and to receive sidetone



II - ARCHITECTURE DESCRIPTION (continued)

Figure 6 : Transmit Compressor Range G_{1max} - G_{1min} = 16.5dB

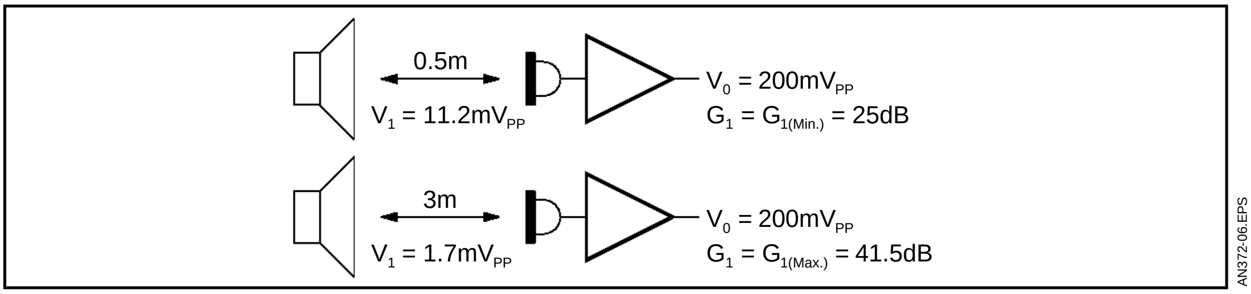
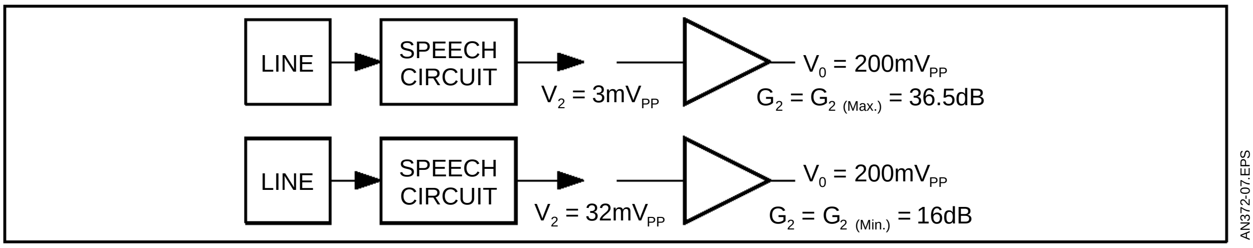


Figure 7 : Receive Compressor Range G_{2max} - G_{2min} = 20.5dB

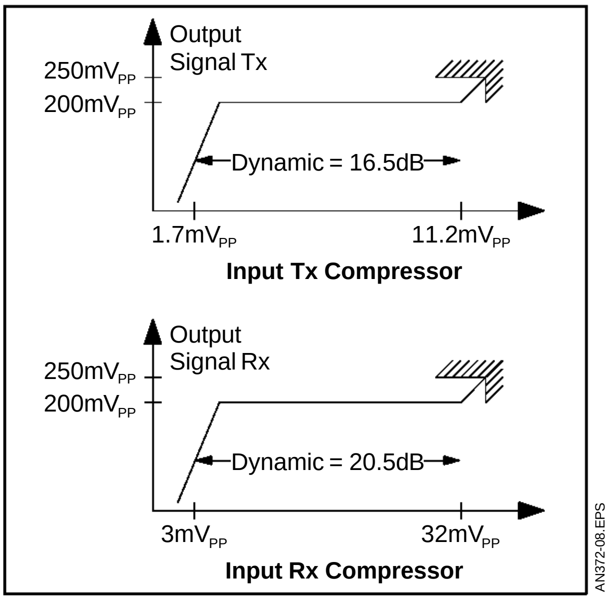


II.5.3 - Self-adaptative Compression with TEA7540

The compressors in the inactive (dominated) channel works at a fixed gain which can never increase but may decrease following an increase of dominated channel signal.

a) Behaviour of a dominant channel compressors : see Figure 8

Figure 8 : Compressor Gain versus Input Signal

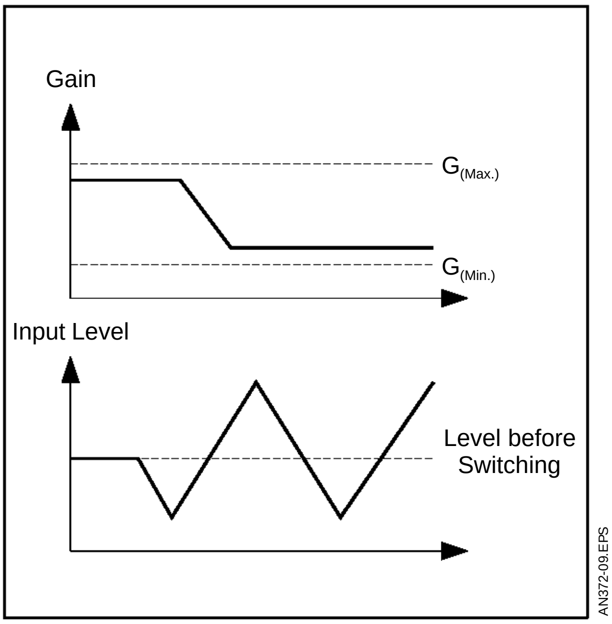


b) Behaviour of a dominated channel compressors : see Figure 9.

As shown in Figure 9 when the Rx (or Tx) compressor input voltage decreases the gain is always held at its previous value.

When this voltage increases to a value higher than previous value, the gain decreases.

Figure 9



II - ARCHITECTURE DESCRIPTION (continued)

II.5.4 - Detection of Typical Noise Condition with TEA7540 (Figure 10)

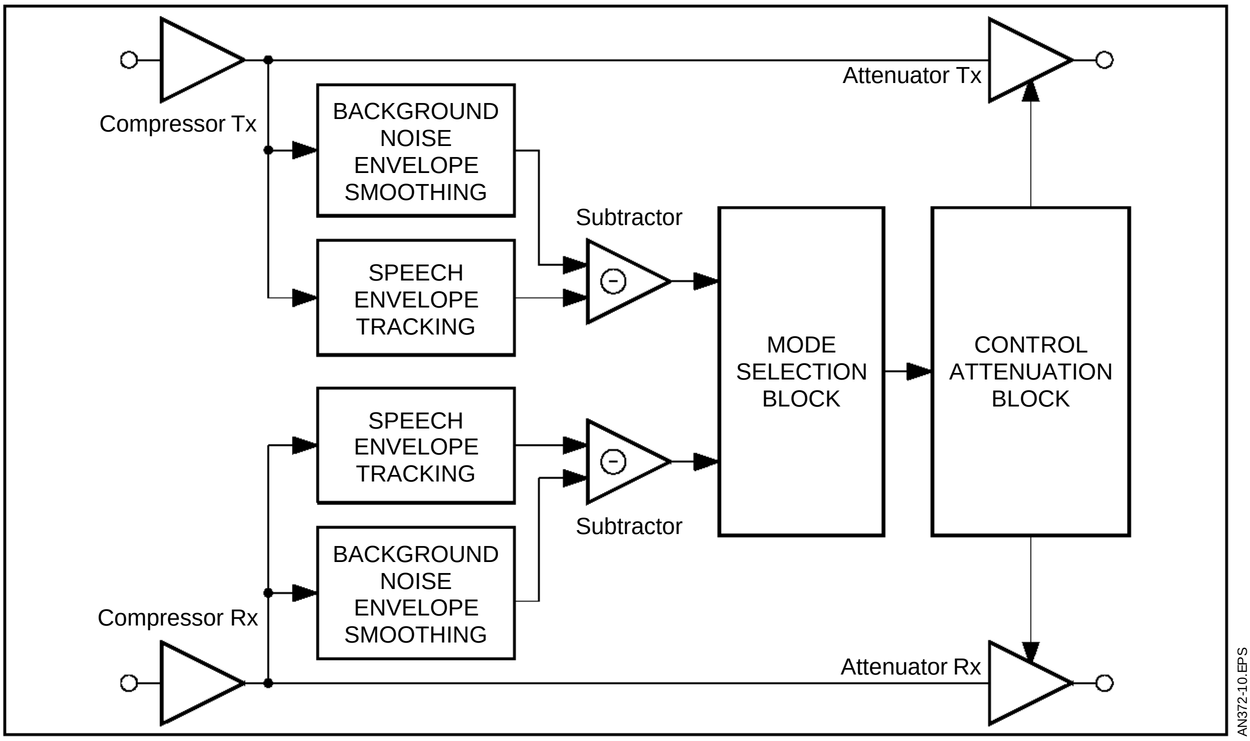
TEA7540 is able to :

- **recognize** the presence of a speech over a wide range of noise levels in both transmit and receive channels.
- **select** the appropriate mode, transmit, receive or idle, by extracting and comparing the burst signal component (speech), of both the Tx and Rx channels, while rejecting the long-term steady-state noise components.
- **switch**, such as appropriate mode, in the transmit, receive or idle. This involves appropriate switching of Tx and Rx channel attenuation values.

II.5.5 - Flexible Application

- **All** the different time constants are externally programmable.
- **Attenuation** values are externally programmable.
- **The** same application can be used in monitoring mode (group listening) via the chip select pin.

Figure 10 : Background Noise/Speech Discrimination



III - LOOP GAIN PRINCIPLE (see also section V.2)

III.1 - The Voice-Path Loop-Gain Calculation

The hands-free telephone circuit contains a loop comprising the transmit and receive path amplifiers and attenuators, in association with sidetone and acoustic coupling attenuation (Figure 4). The loop-gain, LG, must be lower than 0dB to avoid instability.

Loop Gain = LG = G1 - AT +GT - A1 + GR + G2 - AR + GL - A2 (dB) <0dB

The compressor gains, G1 and G2, may be expressed in the form:
G1 = G1max - dG1 (dB)
G2 = G2max - dG2 (dB)

In these equations, G1max and G2max are constant gain values, and dG1 and dG2 represent effectively attenuations introduced by the compressor amplifiers to handle signals larger than minimum.

$$0 < dG1 < 16.5dB$$
$$0 < dG2 < 20.5dB$$

In the basic loop-gain equation AT represents the attenuation introduced in the transmission channel when the circuit is in either the receive or idle modes. Similarly AR represents the attenuation introduced in the receive channel when the circuit is in the transmit or idle modes.

Note : in receive mode AR = 0 in transmit mode AT = 0 in idle mode the total attenuation is equally shared between two channels.

Conceptually, the circuit of Figure 4 may be redrawn as shown in Figure 11 where G1max, GT, GR, G2max, and GL represent amplifiers of fixed gain, and the dynamic components dG1, AT, dG2, and AR consitute an attenuator block (AS).

The loop-gain equation may now be written in the form :
LG = (G1max+G2max+GT+GR+GL)-(A1+A2)-(AT+AR+dG1+dG2)

Hence : Loop Gain = Const(gain) - Const(att) - (AT+AR+dG1+dG2)

or, alternatively :
Loop Gain = Constant - (AT+AR+dG1+dG2)

Let the total controlled attenuation be represented by : AS = AT + AR for dG1 and dG2 #0

Loop Gain = LG = Constant - (AS+dG1+dG2) ..(1)

The total dynamic attenuation, AS, is shared between two controlled attenuators, one each in the receive and transmit channels, according to the mode of operation.

Let AS0 = AT0 +AR0 for dG1 = dG2 = 0, i.e. for maximum compressor gains. AS0 is the total attenuation inserted by the attenuation control blocks in the two paths in function of the operating mode, in particular :

	Tx Mode	Rx Mode	Idle Mode
AT0	0dB	AS0	AS0/2
AR0	AS0	0dB	AS0/2

The AS0 value is programmed by an external resistor between 10dB and 60dB as Table 1 shows. The AS0 value must be choosen in order to guarantee the stability condition (LG0 < 0dB) for all the operating modes and with all the signal level conditions. It must compensate the sum of the electrical and acoustical coupling and the constant gains. Let LG0 the corresponding value of the loop gain.

Then :
Loop Gain = LG0 = Constant - AS0 << 0dB ...... (2)

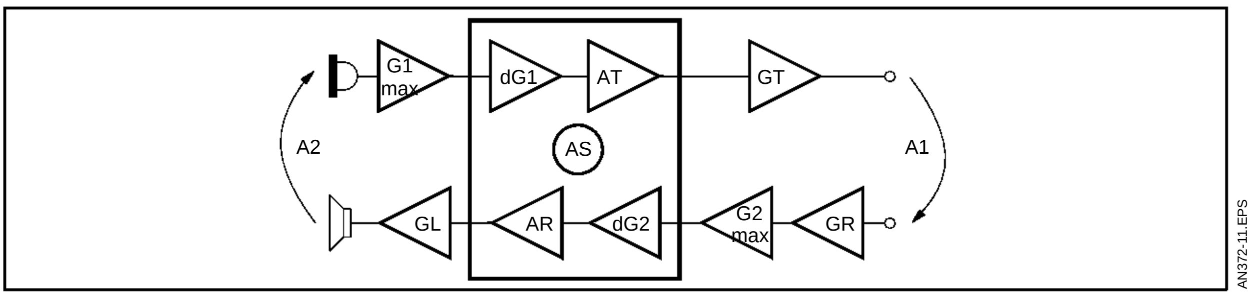
For any other compressor gain/operating mode conditions we may write, using the expression (1) and (2) :
LG - LG0 = (AS0-AS) - (dG1-dG2) ....(3)

When the AS0 value has been choosen TEA7540 will control the value of AS = AT + AR so that LG - LG0 = 0 for all operative conditions of compressor gain (i.e. all values of dG1 and dG2), and all modes of operation (i.e. all possible modes of sharing the total controlled attenuation, AS, between the two paths, in this way the system will approach to a full duplex behaviour because TEA7540 will keep always LG at the max value guaranting the stability.

from rule 3 we have : AS = AS0 - dG1 - dG2

AS is a parameter time varying in function of level signals and shared between receive and transmit paths in function of the operating modes.

Figure 11 : Attenuations Block Principle



III - LOOP GAIN PRINCIPLE (see also section V.2) (continued)

This provides that LG is always as close as possible to LG0 value.

Thus we have:

a) In transmit mode we have zero attenuation in the transmit channel, AT = 0 and hence :
AR = AS0 - dG1 - dG2

Note : Remember that the gain of the maximum receiving compressor (G2) is frozen at its previous value before the switching, and dG2 varyes as explained in section II.5.3 whereas dG1 is varying according to input microphone signal. Consequently when the system is in Tx mode and the level of microphone is increasing the compressor gain decreases. TEA7540 will decrease AS on the receive path by an equal amount.

b) In receive mode we have zero attenuation in the Receive channel, AR =0 and hence :
AT = AS0 - dG1 - dG2

Note : Remember that the gain of the maximum transmitting compressor (G1) is frozen at its previous value before the switching, and dG1 varyes as explained in section II.5.3 whereas dG2 is varying according to the Tx input signal from the line. In the same previous way when the system is in Rx mode and the level on the line signal is increasing the compressor gain decreases. TEA7540 will decrease AS on the transmitting path by an equal amount.

c) In idle mode the attenuation is shared equally between the two channels and hence :
AT = AR = AS/2 = (AS0 - dG1 - dG2)/2
where dG1 and dG2 are frozen to them previous values.

In TEA7540 the attenuator controller processes the compressor signals (which provides a measure of the values of dG1 and dG2) and produces control voltages which set the attenuation levels AT (Receive mode), AR (Transmit mode), or both AT and AR (Idle mode) so as to ensure stability under all conditions of operation.

Table 1 : Table of Programmable Attenuation AS0

R Pin 24	Programmable Attenuation AS0	R Pin 24	Programmable Attenuation AS0
0K	10.4dB	8K	40.2dB
1K	13.2dB	9K	44.3dB
2K	17.0dB	10K	48.0dB
3K	20.3dB	11K	52.0dB
4K	24.3dB	12K	55.0dB
5K	28.4dB	13K	57.7dB
6K	32.5dB	14K	60.2dB
7K	36.4dB	15K	62.4dB

with G1, G2 = max and dG1, dG2 = 0

III.2 - Examples

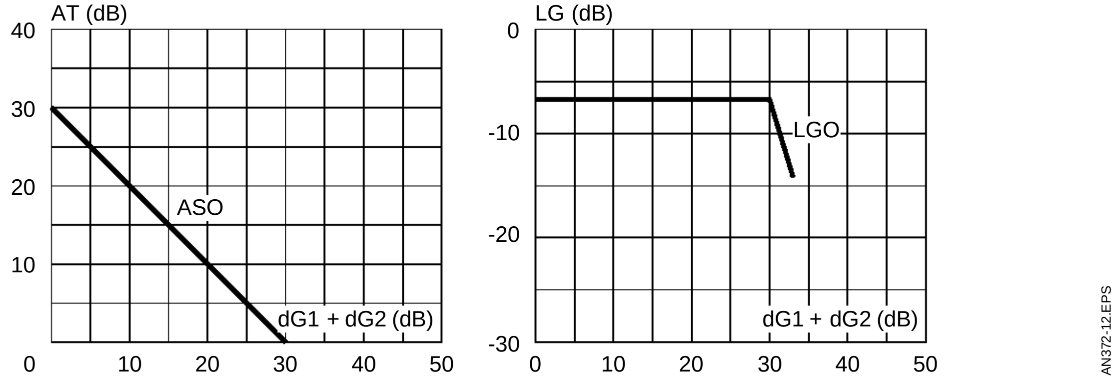
In the following examples (Figures 12, 13 and 14) the circuit is in receive mode with required total attenuation values of AS0 = 30dB, AS0 = 37dB, and AS0 = 50dB (depending on different values of the constant).

A resistance value of approximately 5.5kΩ connected to Pin 24 yields a maximum attenuation of 30dB with both compressors operating under maximum gain conditions. With the compressor gains reduced by a total of dG1 + dG2 = 30dB the attenuation AT is reduced to its minimum possible value of 0dB. With any further decrease of compressor gains the loop gain is reduced by up to 7dB (see Figure 12).

In this case (Figure 13) the value of AS0 is set to a value of 37dB (resistance value of 7.2kΩ connected to Pin 24) with both compressors operating at maximum gain. The attenuation is reduced by 1dB for each decrease of 1dB in total compressor gain up to the maximum of 37dB.

In this case (Figure 14) the value of AS0 corresponding to the maximum compressor gain setting is adjusted to 50dB (using a value of about 10.5kΩ connected to Pin 24). Although the available dynamic range of attenuation values is 52dB the actual attenuation range is limited to 37dB.

Figure 12 : Example 1. AS0 = 30dB with the const. = 24dB



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III - LOOP GAIN PRINCIPLE (see also section V.2) (continued)

Figure 13 : Example 2. AS0 = 37dB with the const. = 31dB

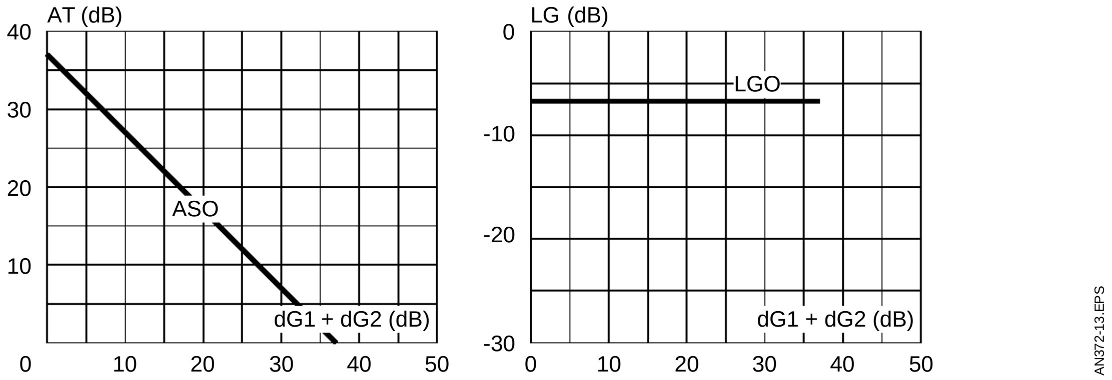
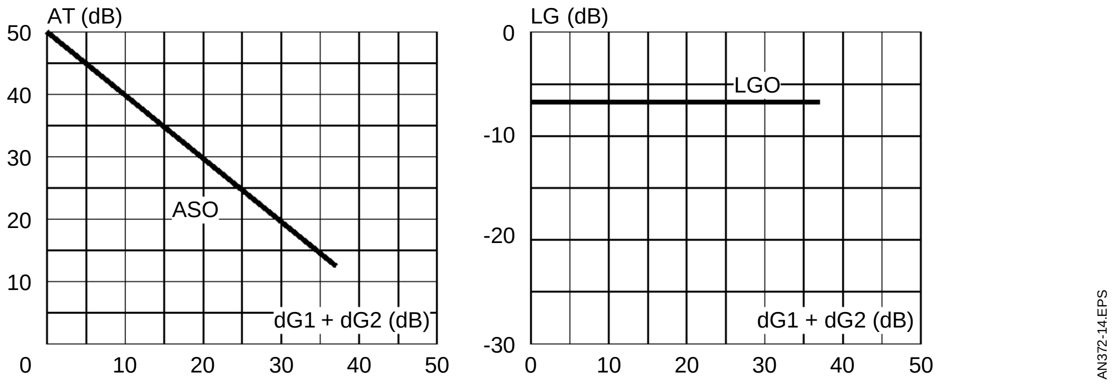


Figure 14 : Example 3. AS0 = 50dB with the const. = 44dB



III.3 - Volume Control

The receive channel contains a volume control which is active only in receive mode. TEA7540 attenuator control unit adjusts the attenuations in the transmit and receive channels so that the overall loop gain remains constant. In the receive mode any additional attenuation AVOL added (or substracted) in the Rx channel is compensated by an equal attenuation substracted (or added) in Tx channel.

In the transmit mode, however, the attenuation AT remains fixed at 0dB and variations in AR are hence not compensated. The loop gain remains < 0dB because the stability (see rules in section III.1) is guaranted with the volume control attenuator set to the minimum value (AVOL = 0dB). In the receive mode the rules for AT and AR become :

AR = 0dB + AVOL
AT = AS - AVOL

An external potentiometer connected to the Pin 23 fixes the value of the attenuation of the volume control circuit (AVOL) in the range from 0dB to 33dB.

III - LOOP GAIN PRINCIPLE (see also section V.2) (continued)

III.4 - Tx/Rx Switching Control

III.4.1 - The IDLE/Transmit/Receive Comparator Principle (see section IV.1.3)

An important functional requirement for a Handsfree telephone system relates to the switching between idle, transmit, and receive modes.

This switching should respond rapidly to the sudden increase of signal which occurs with the onset of speech, without reacting improperly to the brief periods of interword silence which occur in normal connected speech.

The associated control is effected by comparing the amplitudes of the signals produced by the two compressor/amplifier units for the receive and transmit channels. For each channel, two signals are derived, the first being proportional to the peak amplitude, and the second to the mean-squared amplitude, of the signal produced at the compressor output.

The peak amplitude signal, VPEAK, produced by a fast-acting (short time-constant) detector circuit gives a measurement of the speech signal with its burst nature. The mean-squared amplitude signal, V_{NOISE}, produced by a long time-constant circuit, gives a measure of the noise signal which is of an essentially constant nature.

The role of the comparator circuit is to recognize the channel in which a burst-type (speech) signal first occurs. A hysteresis component (attenuator) is then switched into the opposite channel to effectively raise the threshold of the quiscent (dominated) channel with respect to the active (dominating) channel.

III.4.2 - The Selection of The Hysteresis Value

Simplifying Assumptions

a) We will consider the case where the noise components in both channels are zero.
   In practice the comparison circuit works with respect to the difference signal, V_{PEAK} - V_{NOISE}.
   To cater for this case, the Hysteresis value may be increased beyond 6dB by external means at the disposal of the designer as appropriate to the needs of his particular application.
   This is achieved by connecting external resistances between Pin 6 and Earth (Rx channel hysteresis).
   The user has been offered this option as a preferable alternative to a built-in larger internal hysteresis, since it allows him to tailor circuit performance closely to the needs of his particular application.
   By way of example, this facility enables a quasi full-duplex mode to be realized since both the overall loop attenuation and also the individual hysteresis values may be held at the minimum possible values.
   The completely individual flexibility offered by the SGS-THOMSON chip-set enables the designer to produce economical, viable designs to cover a wide number of cases, e.g. analog telephone systems with loop current power supplies, digital telephone systems with extremely high sidetone rejection and high loudspeaker efficiencies, or long-line systems with low loudspeaker efficiency. All of these examples may be satisfactorily "tuned" by external adjustment of the Hysteresis values.
   The value of the external resistor (R5 or R6) required to fine tune the hysteresis value for any particular application may be chosen using the following table as a guide.

$$HYT = 6dB + H_{OT},\ HYR = 6dB + H_{OR}$$

R Pin 5 - R Pin 6	H_{OT} - H_{OR}
1M	0dB
14K	3dB
5.6K	6dB
3.3K	9dB
2.3K	12dB

b) because of the essential symmetry of the Tx and Rx channels we will consider only the case of the Tx dominant mode, identical conditions of operation occuring for the Rx dominant mode.

1 - Normal Operating Mode (Figure 15)

In the normal operational mode the input signal in the dominating channel (Tx) is assumed to lie within the dynamic operating range of the compressor amplifier, and hence the compressor output voltage is 100mV, independently of the actual signal input.

In this case with compressor outputs in both channels of 100mV, a Hysteresis value of 6dB in the dominated channel (Rx) will produce a good margin undesired speech chopping produced by improperly mode switches.

Accordingly SGS-THOMSON has placed a fixed value of 6dB in the circuit.

2 - Abnormal Range of Operation (Figure 16)

This case relates to the situation where the input signal is beyond (below) the dynamic operating range of the compressor and the Tx compressor output is less than the normal value of 100mV. This may arise in two cases :

a) There is relatively large separation between the sender's mouth and the Handsfree system microphone.
b) In the intersyllabe and interword periods of silence in normal connected speech.

In a typical example, with a dominant channel output of 50mV a hysteresis value of 6dB may be not adequate.

III - LOOP GAIN PRINCIPLE (see also section V.2) (continued)

Figure 15

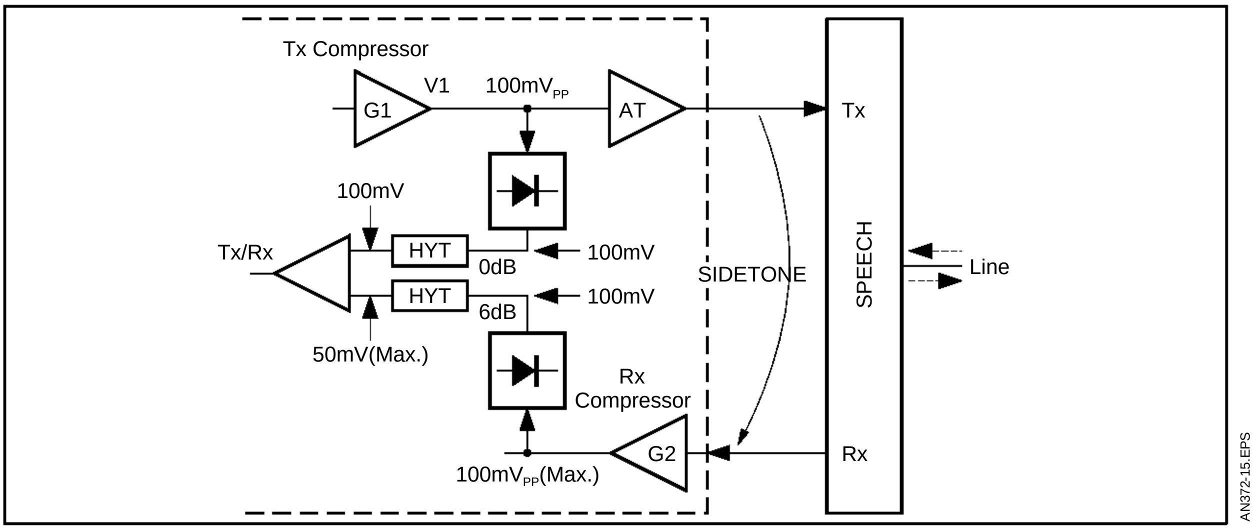
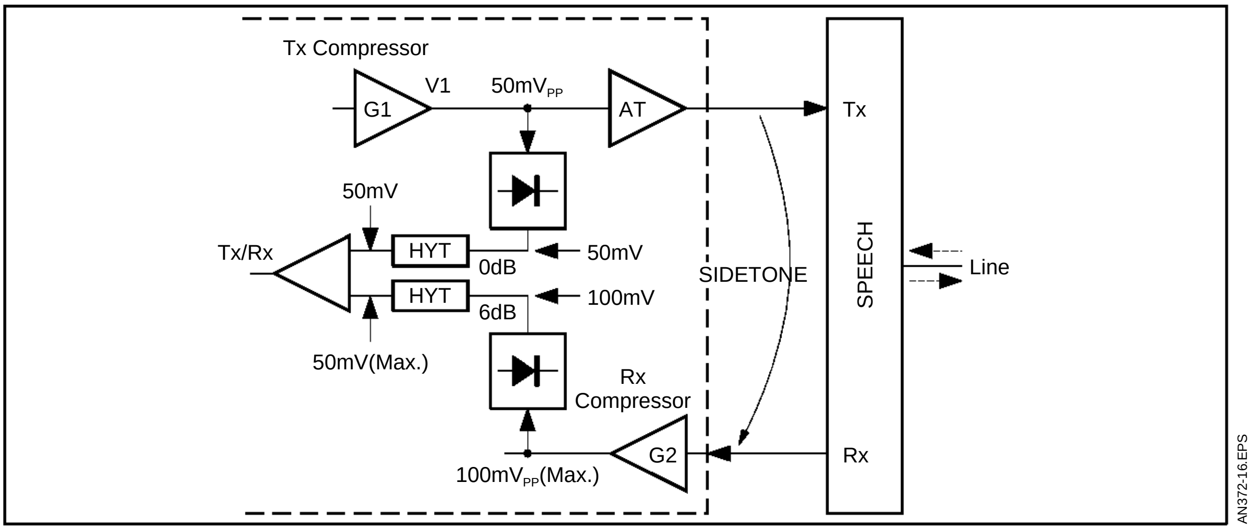


Figure 16



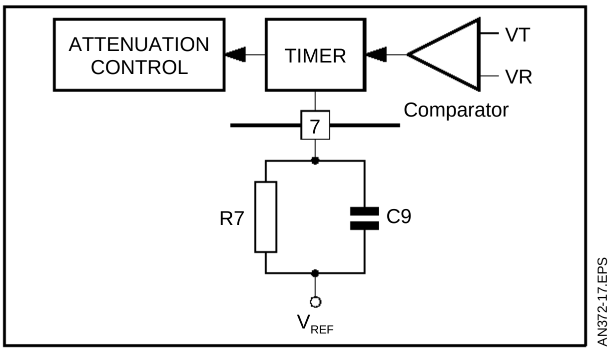
III.5 - Timer

The timer in Figure 17 is designed to allow rapid transitions to the Transmit or Receive modes from any operating mode (Transmit/Receive/Idle), and relatively slow transitions to the Idle mode from either transmit or receive mode.

The rapid transition to one of the active modes, transmit or receive, is required in order to avoid the loss of the beginning of a segment of speech. This is accomplished by the timer providing, as appropriate, a charge/discharge current to the external capacitor (t = CV/I).

The slow transition is required to avoid switch in the idle mode in the short periods of silence between individual spoken words. In the Idle mode the timer provides no charge/discharge current, and the external capacitor is charged/discharged via the external resistor with an effective time constant of about 1 second (t = 2.2 RC).

Figure 17 : Timer Simplified Schematic



IV - CIRCUIT DESCRIPTION

IV.1 - Main Functional Blocks

IV.1.1 - Signal Compressors

The signal compressors are managed by the circuitry shown in Figure 18.

The microphone input signal plus the transmit channel input noise are amplified by a variable gain AGC amplifier (labelled compressor in Figure 16) to yield an output signal with a mean envelope amplitude of $100mV_{PP}$. The peak detector and noise/speech comparator are used to compare the noise/speech amplitude to a reference DC voltage of 100mV, and to generate a difference voltage which controls the switch labelled K in Figure 16. The position of switch K controls the discharge or charge of external capacitor C10 by currents of 100µA or 1.25µA respectively. The integrated voltage developed across C10 is amplified by the V/I converter and produces an AGC voltage to rapidly (typically within 22ms) decrease the compressor gain, and to slowly (typically within 1.8s) increase the compressor gain (see section IV.1).

The AGC system has hence a dynamic response which reacts quickly to the increase of amplitude at the beginning of each word, but decays slowly at the end of a word. This feature prevents the system from increasing the gain, and hence noise-level, in the periods of silence between successive words in a passage of connected speech. Without this feature, the variations of noise level which could occur within a passage of connected speech would be extremely disagreeable to the listener.

The behaviour of a disabled compressor is describe in section II.5.3.

The compressor of inactive part holds its previous gain and the capacitor (C9 in Tx path, C10 in Rx path) is disconnected from the current sources. The voltage decreasing (Pin 8 Tx or Pin 20 Rx) depends on the quality of the capacitor. Use a capacitor with as low leakage current as possible.

IV.1.2 - Peak Detector and Noise Monitors

The circuit of Figure 19 shows the method of generation of two signals, VTx and VNTx (or VRx and VNRx). Signal VTx represents the envelope amplitude of the combined noise and useful signal at the compressor output, whereas VNTx represents the envelope amplitude of the noise component only. Hence if VTx = VNTx the compressor output is a noise output solely (no transmit signal present). However, if VTx > VNTx the transmit channel compressor output contains a useful signal component, either speech or dialling tone.

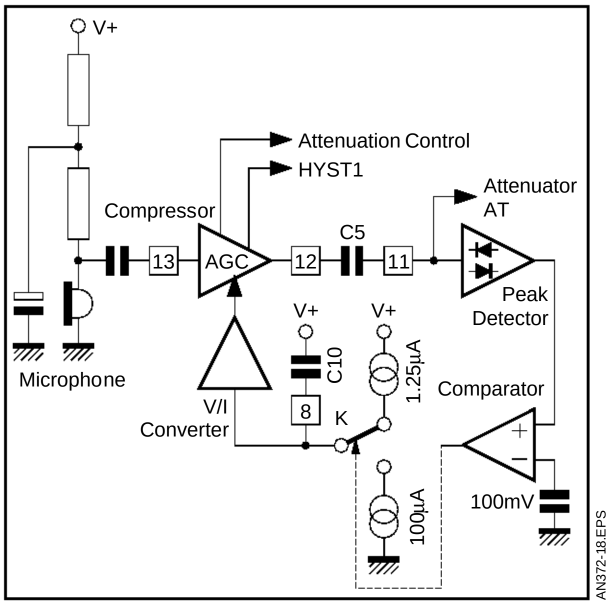
Output VTx is equal to the voltage across capacitor C12 which may either be charged rapidly by a current of 20µA if transistor T1 is cut off, or discharged slowly via R1 when transistor T1 is driven into saturation. The output of comparator amplifier A1 is used to drive T1 into cut-off when the instantaneous compressor output is greater than VTx and drives T1 into saturation when the compressor output falls below the value of VTx. Hence VTx is a voltage that rapidly follows any increase of the compressor's noise plus signal output and follows any decrease of that output with an time constant fixed by the external R1 C1 values (typically 100ms).

The signal VTx will hence contain a long-term steady-state component equivalent to background noise, plus short-term transient (burst) components which follows the variations of amplitude due to speech signals in the compressor output. These latest components may be eliminated by the similar circuit comprising T2, A2, and timing components R8 and C11. These timing components define a much longer time-constant of approximately 10s resulting in an output VNTx which effectively contains only the long-term components of the compressor amplifier output.

The additional components comprising the comparator amplifier A3 and transistor T3 limit the maximum value of VNTx to 36mV. This feature enables the circuit to identify the case where the useful signal is a dialling tone. In this case VNTx will be limited to a value of 36mV whereas VTx will increase to a greater value.

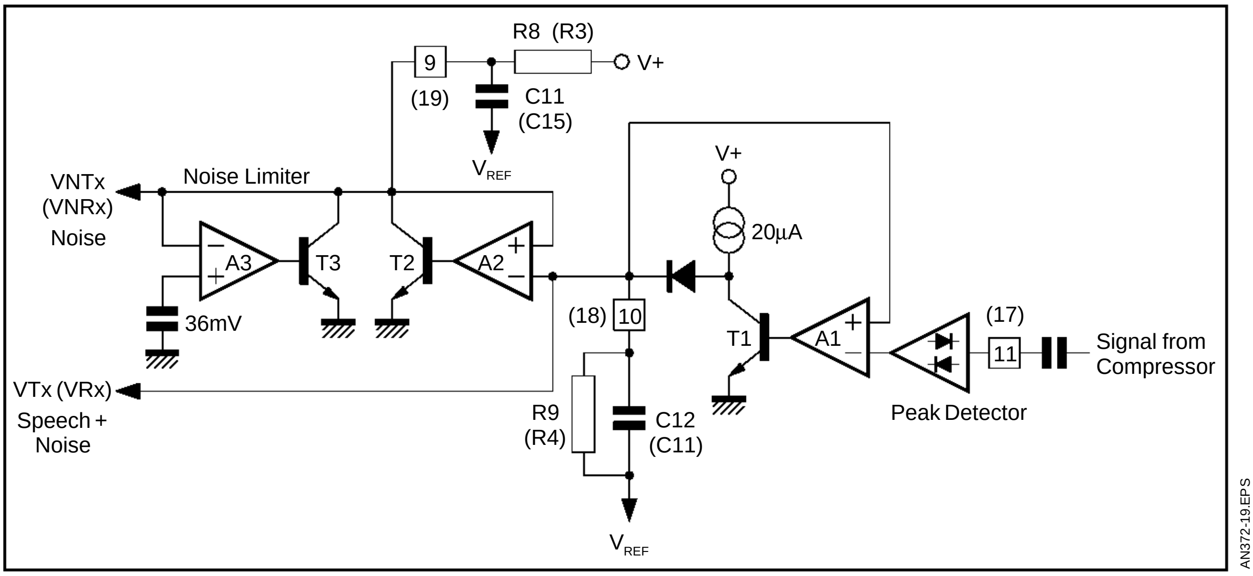
The amplitude relationship, VTx > VNTx, corresponding to the presence of a useful signal component in the compressor output, is hence satisfied in this case also.

Figure 18 : Signal Compressors



IV - CIRCUIT DESCRIPTION (continued)

Figure 19 : Peak Detector and Noise Monitors (see also section IV.3.3)



IV.1.3 - Comparators

The comparator circuits in Figure 20 determine the specific modes of operation : SPEECH/NOISE and Transmit/Receive.

If VTx exceeds VTNx by more than 8mV, a useful signal component is considered to be present in the Transmit channel compressor output.

Similary if VRx exceeds VNRx by the same margin a signal component is present in the receive channel.

If either or both of these conditions are true the OR gate output is driven high (positive logic TRUE condition) and the SPEECH/NOISE (i.e. SPEECH/NOISE mode) output is set to 1.

Idle mode is set when both VSTx and VSRx are less than 8mV.

The transmit/receive mode signal output is determined by a comparison of the transmit channel signal VSTx = VTx - VNTx with the receive channel signal VSRx = VRx - VNRx.

Before this comparison the voltage representing the useful signal in the dominated channel is attenuated by either HYT (transmit channel) or HYR (receive channel), whereas the signal in the dominating channel is passed to the comparator input without attenuation (i.e. HYT = 0dB for Tx dominant ; HYR = 0dB for Rx dominant).

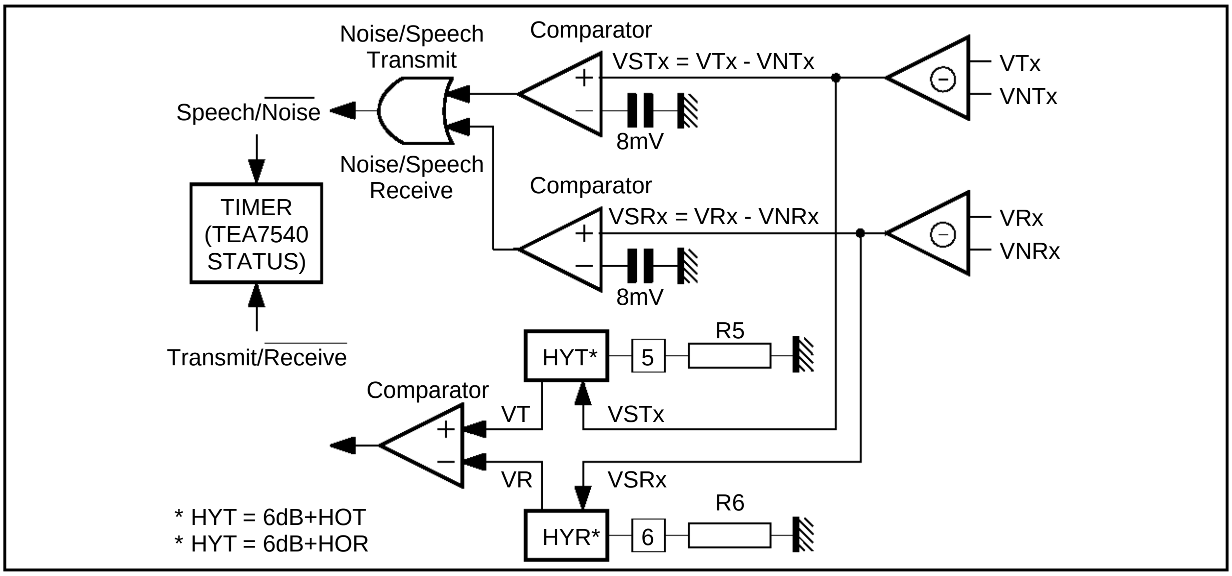
The status of TEA7540 IC's can be resume by the following true table :

Noise/Speech Transmit Comparator	1	X	1	0	0
Noise/Speech Receive Comparator	X	1	0	1	0
Transmit/Receive Comparator	1	0	1	0	X
TEA7540 Status	T	R	T	R	I

T : Transmit
R : Receive
I : Idle

IV - CIRCUIT DESCRIPTION (continued)

Figure 20



IV.1.4 - Attenuation Control

The organization of the Attenuation control circuitry is shown in Figure 21. As described in sections III.1 and III.2.

the attenuation is required to control the transmit and receive channel attenuators, A_T and A_R , in function of operating mode, the compressor amplifier gains, and the Receive channel volume control setting.

In transmit mode we have zero attenuation in the Transmit channel, $A_T = 0$, and hence :

$$A_R = AS0 - dG1 - dG2$$

In Receive mode we have zero attenuation in the Receive channel, $A_R = 0$, and hence :

$$A_T = AS0 - dG1 - dG2$$

In Idle mode, the attenuation is shared equally between the two channels and hence :

$$A_R = A_T = AS/2 = (AS0 - dG1 - dG2)/2$$

The controller block in Figure 21 develops the required outputs to control A_T and A_R, in terms of the AGC inputs dG1 and dG2 (which depend on the compressor amplifier gains), the mode control signals, the receive channel volume control setting, the maximum value of total attenuation AS0 (determined by the external resistor) and the Timer output voltage, V_{TIM}.

The timer is required to control the speed of transition between idle, transmit, and receive modes. The transition into transmit or receive (i.e. speech) mode must be quick enough in order not to miss the beginning of a segment of speech. Transition from transmit or receive mode to the Idle mode must be slow in order to avoid an increase of noise level between successive words of a continuous passage of speech.

These changes have an appropriate time constant by discharging the capacitor C9 by acurrent of 50µA or by charging the same capacitor through the resistor R7 in Figure 21. This selection is controlled by the timer circuit by setting the switch K.

The timing diagram of Figure 22 illustrates the dynamic response of the attenuator values, A_T and A_R, in the two cases of a transition between Rx and Tx and of transition from Rx mode to Idle mode.

IV.1.5 - Noise Threshold Hysteresis

Figure 23 shows the noise threshold hysteresis circuit which affects the dynamic response for changes from an active mode to the Idle mode (the parameters of this circuit are definied internally and are not programmable by external components.

This circuit inserts additional attenuation into either VSTx or VSRx, according to the appropriate operating mode.

This attenuation acts in conjunction with the timer signals to slightly delay any change from an active to the idle mode and leads to an optimization of the dynamic response of the system.

IV - CIRCUIT DESCRIPTION (continued)

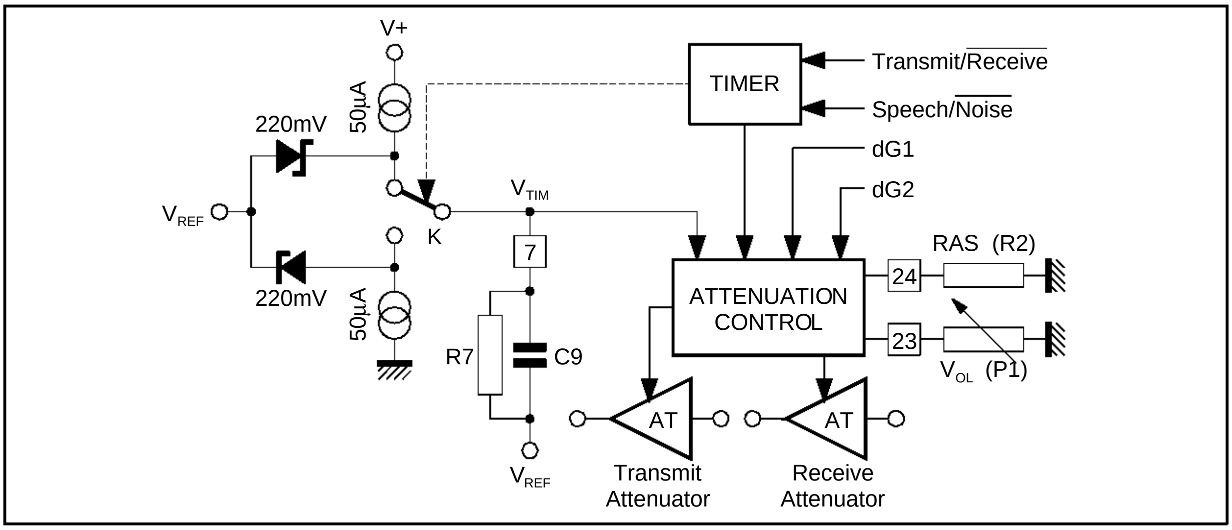


Figure 21 : Attenuation Control

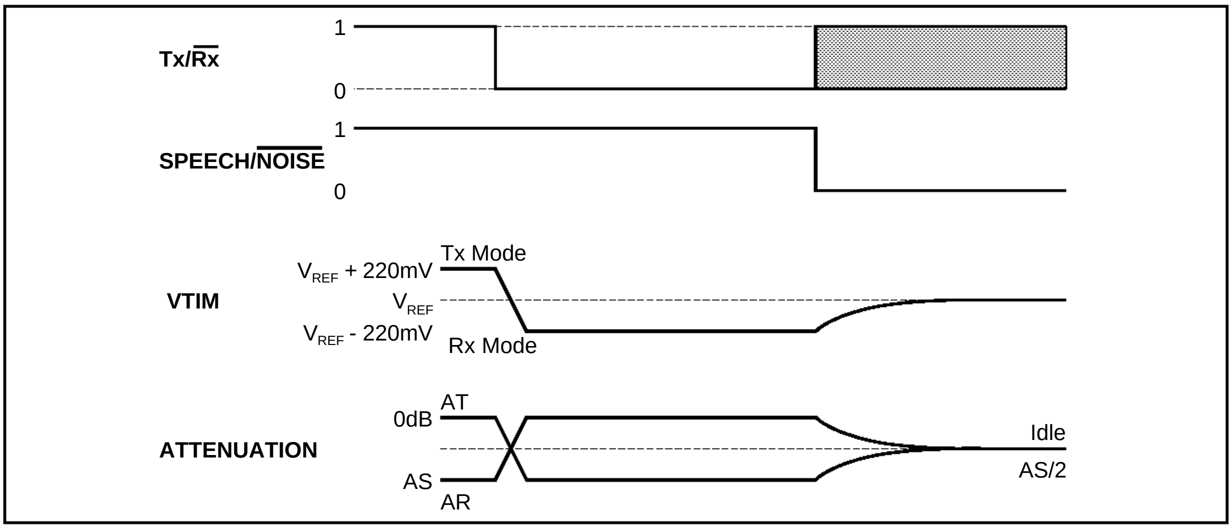


Figure 22 : Transmition Diagram among Tx-Rx, IDLE mode (see also section IV.3)

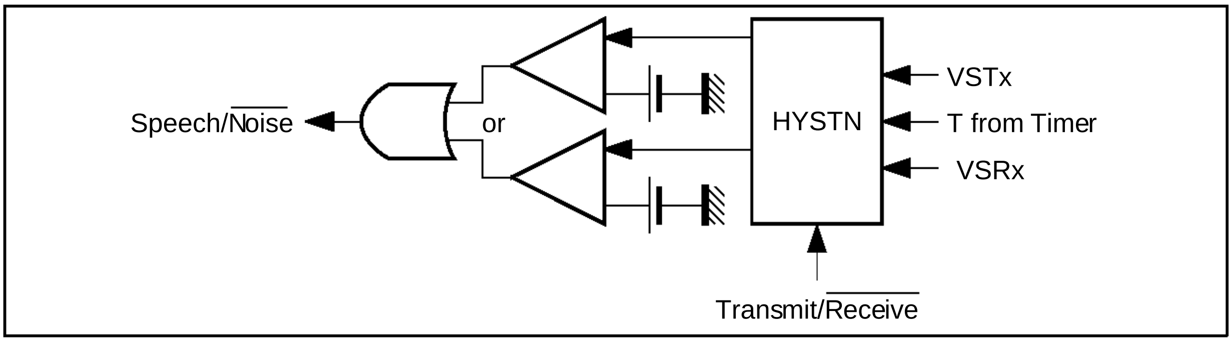


Figure 23 : Noise Threshold Hysteresis

IV - CIRCUIT DESCRIPTION (continued)

IV.1.6 - Chip Select Circuitry

Figure 24 illustrates, conceptually, the functions of the two transmit and the receive channels and their control by means of the external switch CS. There are basically two operating modes :

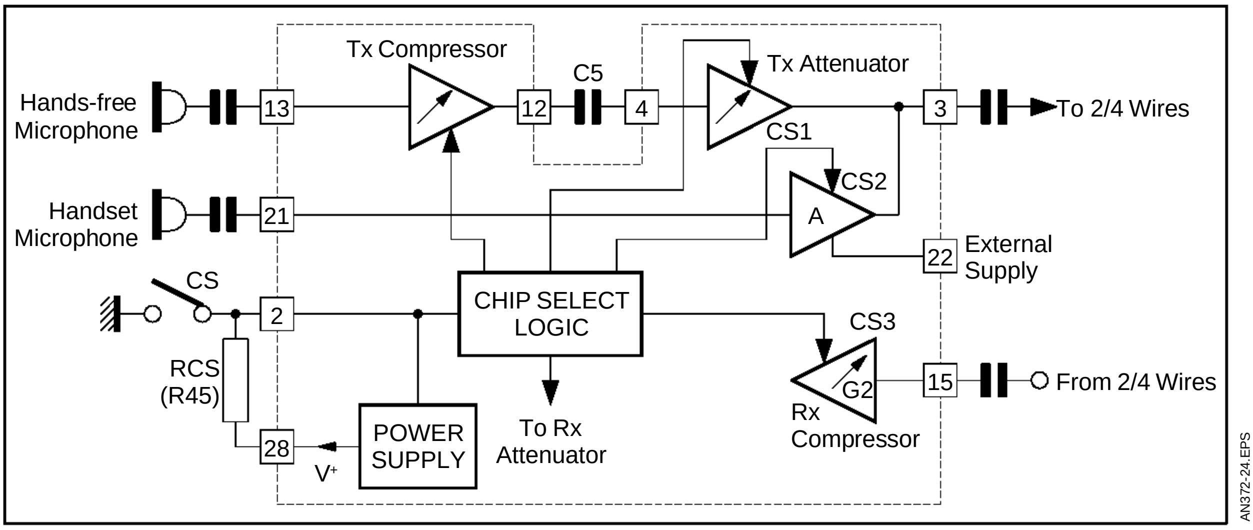
a) CS = ON : Handsfree Mode.
In this mode the handset microphone pre-amplifier is inactive and the two handsfree channels of TEA7540, transmit and receive, are both active.

b) CS = OFF : Monitoring Mode.
In this mode the transmit compressor is inactive and the pre-amplifiers A is active with a fixed gain of 20dB. The voltage of the timer is forced to V_{REF} - 220mV and so the attenuation control block see 0dB on Rx attenuator(A_R).

The DC supply required by the micro-phone pre-amplifier may be provided by a separate power supply, e.g. the one available in the dialling circuit. In this case of the power supply requirements are so small that the pre-amplifier is active, and the handset must be kept active, even under the worst conditions of low available supply current due to long-line connections.

In the monitoring mode the automatic control of the compressor amplifier in the receive channel is inoperative.
The gain of this compressor may however be adjusted by selection of a suitable value of the external resistance R_{CS} (R45) connected between Pin 2 and Pin 28.

Figure 24 : Chip Select Circuitry



IV - CIRCUIT DESCRIPTION (continued)

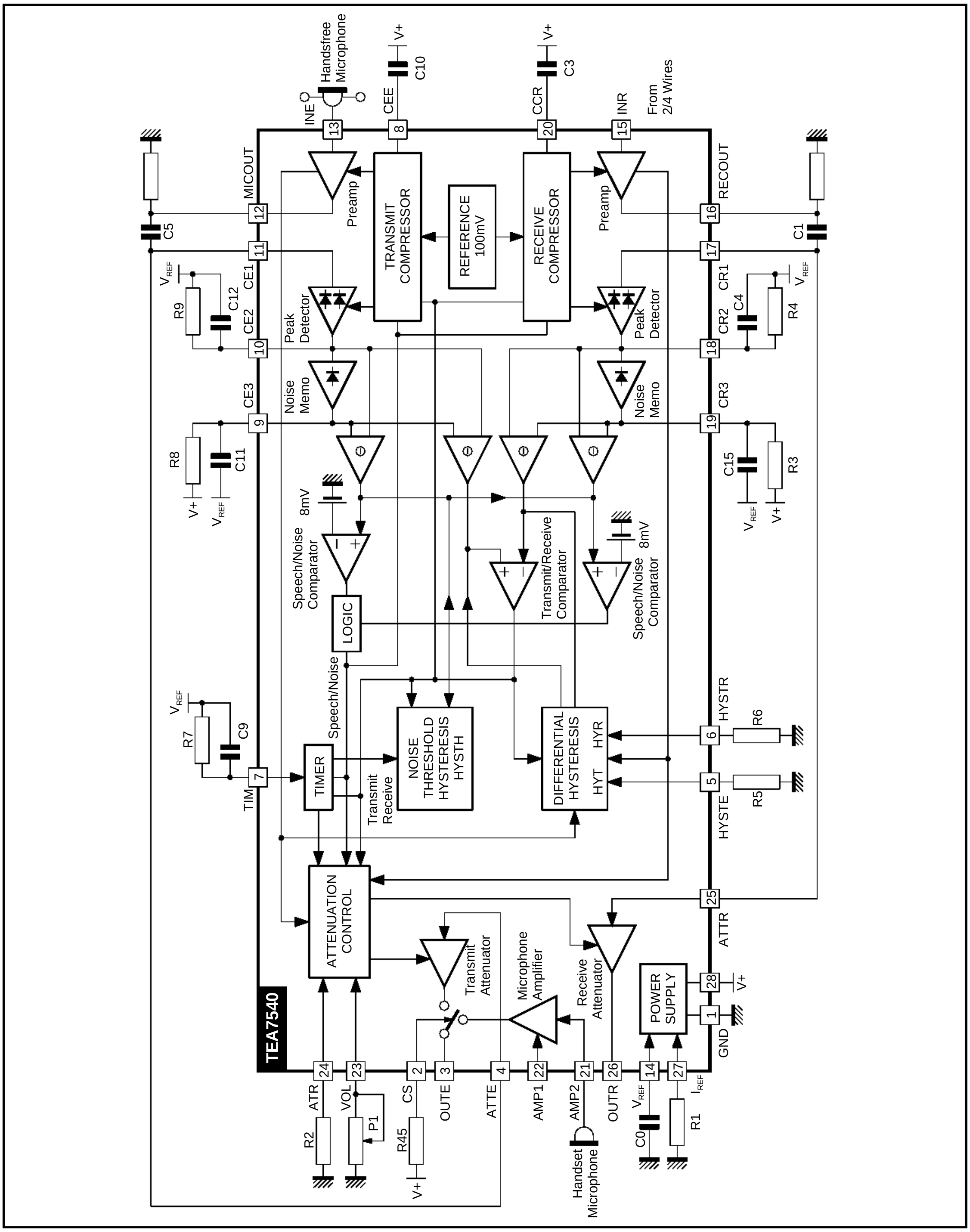
IV.2 - Pin Description

No	Pin	Description
1	GND	Ground
2	CS	Chip Select
3	OUTE	Transmit Attenuator Output
4	ATTE	Transmit Attenuator Input
5	HYSTE	Transmit Hysteresis
6	HYSTR	Receive Hysteresis
7	TIM	Timer
8	CCE	Transmit Compressor Time Constant
9	CE3	Transmit Background Noise Memorization
10	CE2	Transmit Peak Detector
11	CE1	Transmit Rectifier Input
12	MICOUT	Transmit Compressor Output
13	INE	Transmit Compressor Input
14	VREF	Reference Voltage
15	INR	Receive Compressor Input
16	RECOUT	Receive Compressor Output
17	CR1	Receive Rectifier Input
18	CR2	Receive Peak Detector
19	CR3	Receive Background Noise Memorization
20	CCR	Receive Compressor Time Constant
21	AMP2	Microphone Pre-amplifier Input
22	AMP1	Microphone Pre-amplifier Supply
23	VOL	Volume Control
24	ATR	Attenuator Control
25	ATTR	Receive Attenuator Input
26	OUTR	Receive Attenuator Output
27	IREF	Reference Current
28	V+	Power Supply Input

IV - CIRCUIT DESCRIPTION (continued)

IV.3 - Block Diagram

Figure 25



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IV - CIRCUIT DESCRIPTION (continued)

IV.4 - Functional Description of the Pin Allocation

Pin 1 : *Ground*

Pin 2 : *Chip Select (see section IV.1.6)*

There are two possibilities to force the system in monitoring or hands-free modes :

a) By an external switch (see Figure 26)
When the Pin 2 is connected to GND (switch close) the system is set in hands-free mode (V_{PIN2} = 0V).
When the PIN 2 is floating the system is set in monitoring (V_{PIN2} = V_{CC} -400mV). In this way the receive amplifier is set to its maximum value.
It's possible to connect Pin 2 with Pin 28 (supply voltage) by a resistor to program the gain of the receive amplifier how the Figure 26 shows.
This resistor programs an internal bias current (I = 400mV/R) for the amplifier.

b) By external command (µP)(see Figure 26)
It depends on the output stage of the gates.
   1) Open collector
   t_{ON} = Hands-free mode, 0V < V_{PIN2} < V_{REF}
   t_{OFF} = Monitoring mode, V_{PIN2} = V_{CC} - 400mV
   2) Totem pole
   The pin 2 must be connected by an diode in order to not force different source as Fig.
   A to GND = Hands-free mode
   A to 5V = Monitoring mode
   The thresholds are the same of the previous case R programs the amplification the suggested value is > 20kΩ.

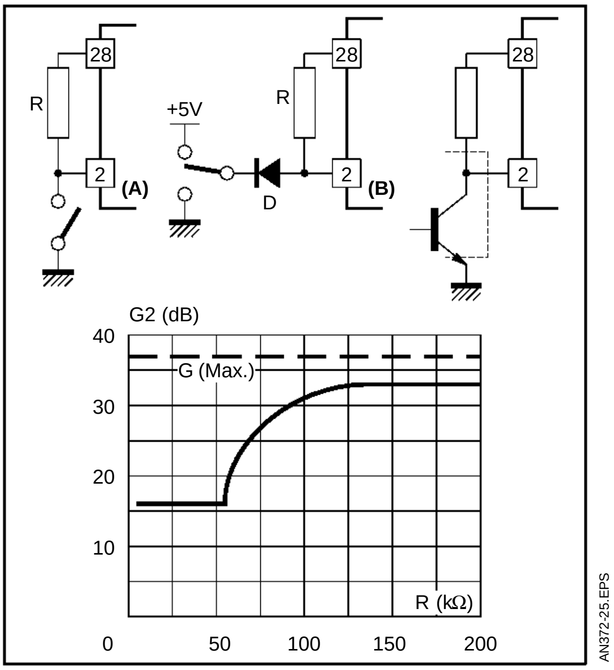
Pin 3 : *Transmit Attenuator Output*

This pin is the output pin for the transmit channel on the hands-free mode (output transmit attenuator) and in monitoring mode (output of transmit amplifier). Output current 15µA.

Pin 4 : *Transmit Attenuator Input*

This pin is normally connected to the transmit compressor output (Pin 12) through a capacitor (CE1) in order to provide a high pass filter. Input impedance 10kΩ.

Figure 26 : Modes by which the monitory mode is forced and the Receive Compressor Gain (G2) programmable by the resistor value connected to Pin 2



Pin 5 : *Transmit Hysteresis (see section III.4.3.b)*

The hysteresis is inserted in the transmit hysteresis block HYT when TEA7540 is in receive mode. It is programmable by an external resistor as in the following table :

R Pin 5	H_{0T}
1.0M	0dB
14.0K	3dB
5.6K	6dB
3.3K	9dB
2.3K	12dB

IV - CIRCUIT DESCRIPTION (continued)

Pin 6 : *Receive Hysteresis (see section III.4.2.a)*

This hysteresis is inserted in the receive hysteresis block HYR when TEA7540 is in transmit mode. It is programmable by an external resistor as in the following table :

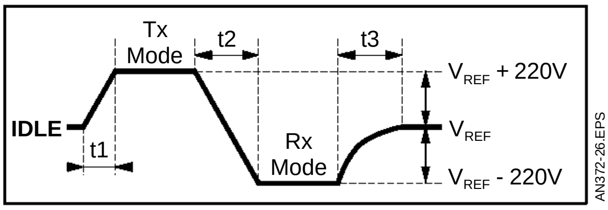
R Pin 6	H_{0T}
1.0M	0dB
14.0K	3dB
5.6K	6dB
3.3K	9dB
2.3K	12dB

Pin 7 : *Timer*

This pin provides information on the state of operation and assumes three values (see section IV.1.4).
- V_{REF} + 220mV for transmit mode,
- V_{REF} - 220mV for receive mode,
- V_{REF} for idle mode.

The switching time constants from Idle to transmit (or receive) are externally programmed by an RC network. The output current at this pin is I = 50µA : see Figure 27.

Figure 27



The voltage transition (V1) at Pin 7 is 220mV from idle to transmit (or receive) and the time constant is t1 (recommended value : t1 = 2.2ms)

The voltage transition (V2) from receive (or transmit) to transmit (or receive) modes is V2 = 2 · V1 the correspondently time constant is t2 = 2 · t1 (recommended value : t2 = 4.4ms).

From transmit (or receive) to Iidle modes is :
This time is longer and controlled by the discharge of capacitor C via the parallel connected resistor.

t3 = 2.2 · R · C (recommended value: t3 = 800ms).

Pin 8 : *CCE Transmit Compressor Time Constant*

The capacitor connected to this pin controls the rate at which the transmit compressor adapts its gain so as to maintain an output voltage level of 200mV peak-to-peak (see section III.4.1).

Therefore the ratio between the time constants is internally fixed at a value of 80. The output current during the increment gain phase I1 = 1.25µA and during the decrement gain phase is I2 = 100µA.

These current sources are used to charge and discharge the external capacitor with the voltage range of 220mV when the incoming signal is decreasing the gain increases with a charge time T1 equal to :

T1 = (V·C)/I1 (recommended value 1.8s)

(where V = 0.22V, I1 = 1.25µA, and C = 10µF)

when the incoming signal is increasing the gain decreases with a discharge time T2 equal to :
T2 = T1/80 (recommended value 22ms)

Pin 9 : *CE3 Transmit Background Noise*

This is used to extract the amplitude of the transmit channel background noise signal. The components connected to this pin determine the transmit channel background noise detector rise time. The maximum voltage drop on the external capacitor is internally clamped at 36mV.

- The rise time is defined by :
  t_R = (36E-3·C·R)/(V_{CC}/2)
  (recommended value : t_R = 9.2s)
- The decay time is the same as the decay time of the peak detector signal (Pin 10) (t_D = 80ms).

Figure 28

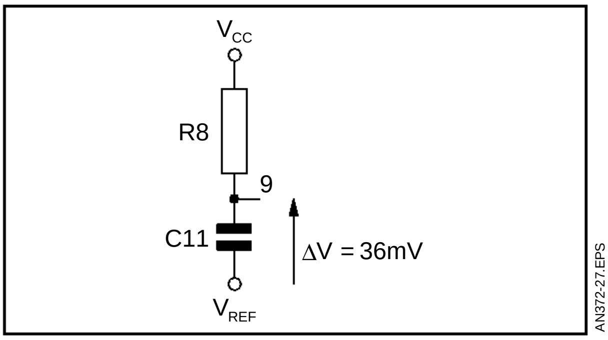
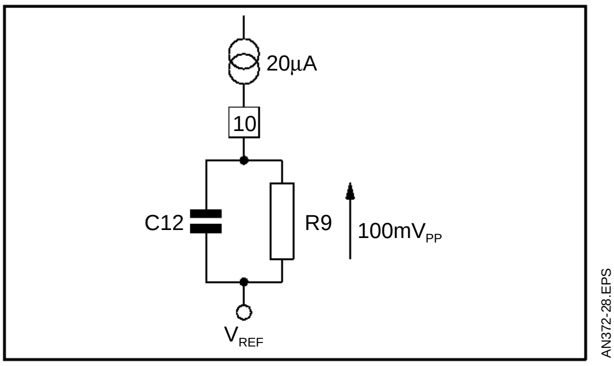


Figure 29



IV - CIRCUIT DESCRIPTION (continued)

Pin 10 : *CE2 Transmit Peak Detector*

- rise time : This is the time required to charge the capacitor C to the maximum value of the amplitude of the compressor output voltage ($100mV_{PP}$).
  $t_R = (C \cdot V)/I = C5 \cdot 10^3$
  (recommended value : $t_R = 2.3ms$)
- decay time : This is the time for the discharge of the capacitor C in the resistor R.
  $t_D = 2.2 \cdot R \cdot C$ (recommended value : $t_D = 80ms$)

Pin 11 : *CE1 Transmit Rectifier Input*

This pin is connected to the Transmit compressor output by a capacitive coupling (typical value: 100nF). The input impedance = 10kΩ.

Note : by connecting in series an external resistor between Pins 12 and 11 it is possible to increase the output voltage of the compressor within the range of its dynamic.

Pin 12 : *MICOUT Transmit Compressor Output*

The maximum output current is 25µA By an external resistor connected between this pin and GND it is possible to increase the threshold of no distorted signal due to :

$$V_{OUT}(mV_{PP}) = \left[25\mu A + \frac{V_{REF}}{R_{EXT}} \right] \cdot 5k$$

Pin 13 : *INE Transmit Compressor Input*

The gain variation is adjusted according to the input signal amplitude

- the gain is between 25 and 41.5dB.
- the compressor output voltage is $200mV_{PP}$.
- the input impedance is 10kΩ.

Hence the input signal voltage range can be calculated to be between $1.7mV_{PP}$ and $11.2mV_{PP}$.

Pin 14 : *Reference Voltage*

$V_{REF} = V_{CC}/2$.
The decoupling capacitor is typically 100µF.

Pin 15 : *INR Receive Compressor Input*

- the gain is between 16dB and 36.5dB.
- the compressor output voltage is $200mV_{PP}$.
- the input impedance is 10kΩ.

Hence the input signal voltage can be calculated to be between 3mV and 32mV.

Pin 16 : *RECOUT Receive Compressor Output*

This output drives the Receive rectifier and the Receive attenuator input. The maximum output current is 25µA. It's possible to increase the no distorted output signal (see Pin 12).

Pin 17 : *CR1 Receive Rectifier Input*

The input impedance is 10kΩ. This input is connected to the Receive compressor output by a capacitor coupling (typical value : 100nF).

Note : by connecting in series a resistor between this pin and Pin 16 it is possible to increase the output level of the compressor within its dynamic (like Pin 11).

Pin 18 : *CR2 Receive Peak Detector*

The rise and decay times are determined by the same rules of the Tx peak detector and depends on :

- a 20mA internal source current,
- an external capacitor,
- an external resistor,
- the compresor output voltage.

The suggested value of rise time is $t_R = 11ms$ and decay time equal to 145ms (C = 22µF, R = 30kΩ).

Pin 19 : *CR3 Receive Background Noise Memorization*

The components connected to this pin determine the Receive background noise rise time. The maximum voltage drop across the external capacitor is also internally clamped to 36mV (see Pin 9).

- the rise time is given by:
  $t_R = (36E\text{-}3 \cdot C \cdot R)/(V_{CC}/2)$
  (recommended value : $t_R = 9.2s$).
- The decay time of this signal is the same as the decay time of the receive peak detector signal ($t_D = 145ms$).

Pin 20 : *CCR Receive Compressor Time Constant Control*

This function is based on the same principle as that explianed previously with respect to Pin 8 for the Transmit Compressor.

Pin 21 : *AMP2 Handset Microphone Pre-amplifier Input*

This pre-amplifier is enabled in the monitoring mode (Pin 2 connected to Pin 28 through an external resistor). It enables the hands-set microphone to be used without an external analog switch. The gain is 20dB so as to be compatible with the required output voltage and the associated input impedance is 50kΩ. The output of the pre-amplifier is internally connected to the transmit attenuator output.

IV - CIRCUIT DESCRIPTION (continued)

Pin 22 : *AMP1 Microphone Pre-amplifier Supply (see section III.1.6)*

This supply has been separated from the main supply (Pin28) to enable it to be supplied by another source (i.e. the same of the dialling circuit) in order to save as much as possible supply current in hands-free mode.

Pin 23 : *VOL (active only in receive mode)*

Using a potentiometer connected to this pin it is possible to introduce a maximum of 32dB of attenuation in the receive channel.

The following table shows the relationship between resistance value and attenuation.

R Pin 23	Attenuation
0K	0dB
1K	3.3dB
2K	6.7dB
3K	10.0dB
5K	16.7dB
10K	33.4dB

Pin 24 : *ATR attenuation AS0*

It should be recalled that this attenuation is fully introduced into the inactive channel in the Speech mode, and that an attenuation of A/2 is introduced into both channels in the noise (Idle) mode.
As discussed previously (see section II.1), it is important, when implementing a high quality Handset design, to ensure that the loop gain is as close as possible to zero dB.
This condition is chosen to avoid the howoling effect otherwise experienced with full-duplex operation.

The following table shows the relationship between the resistance value and attenuation value, AS0.

R Pin 24	Attenuation
0K	10.4dB
1K	13.2dB
2K	17.0dB
3K	20.3dB
4K	24.3dB
5K	28.4dB
6K	32.5dB
7K	36.4dB
8K	40.2dB
9K	44.3dB
10K	48.0dB
11K	52.0dB

Typically, for classical coupling efficiencies, the circuit can operate with a programmed attenuation AS0 equal to 45dB.

Pin 25 : *ATTR Receive Attenuator Input*

This input is normally connected to the Receive compressor output by a capacitive coupling. It is useful to choose this capacitor value to perform a high pass filter with its input impedance (10kΩ) so as to reduce the input noise.

Pin 26 : *OUTR Receive Attenuator Output*

The output voltage is 200 mV_{PP} when the incoming signal is in the compressing range. This pin is connected to the loudspeaker amplifier input. Its maximum output current is 60µA.

Pin 27 : *IREF External Resistor*

The external resistor connected to this pin enables the bias current source of the circuit to be adjusted. A typical value for this component is 3.57kΩ.

Pin 28 : *VCC Positive Supply Voltage*

The minimum working voltage is 2.5V for a current consumption of 2.3mA, and hence line-powered applications can be achieved. This supply can be typically provided by the speech circuit in the same way as the loudspeaker amplifier.

V - INFLUENCE OF THE EXTERNAL COMPONENTS

V.1 - Gain and Dynamic of Tx Compressor

V.1.1 - Gain And Dynamic of Tx Compressor

Gain

- Pin 13 is the output of Tx compressor.
- Pin 12 is the output of the Tx compressor.

The chip is set in the hands-free mode

Note : The figure shows the minimum and the maximum value of the gain. The typical range between values is about 17dB.

Dynamic

This particular behaviour allows to compensate the distance between the speaker and hands-free microphone from 0.5 meters to 3 meters.

We can divide the dynamic curve (Figure 31) in three areas :

1) The first area shows a linear increase of output signal according to the increase of input signal. The AGC of the compressor in this region is not active. The gain is maximum (see also Figure 30).
2) The second area is characterized by a constant output level and in this region the AGC of the compressor is active, its output is kept at 200mV$_{PP}$. The gain is changing from the maximum to the minimum value (see also Figure 30).
3) The third part shows again an increase of the ouput level and in this part the AGC of the compressor is no more active. The gain is now minimum (see also Figure 30).

Note : The figure shows that within 0.9 and 5.7mV$_{RMS}$ the compressor keeps the output level constant and about equal to 220mV$_{PP}$.

If we apply a modulated signal at the input of the microphone we can see the particular behaviour shown in Figure 32.

As the input signal is increasing the compressor output follows it lineary up to the AGC threshold (0.9mV$_{RMS}$), then the output is maintained at constant value (200mV$_{PP}$).
When the input signal is higher than 5mV$_{RMS}$ the output signal starts to increase.
For output signal higher than 250mV$_{PP}$ its distortion increases, by an external resistor between Pin 12 to GND it is possible to increase this thereshold up to 400mV$_{PP}$ (100k).

V.1.2 - Gain and Dynamic Of Rx Compressor

For this analysis we have to consider as the measurement pins :
- Pin15 as input of Rx compressor,
- Pin 26 as output of Rx compressor.

Previous comments are still truth. The only different characteristic is the value of the dynamic range, 20dB in this case.

We can divide the dynamic curve (Figure 34) in three areas :

1) The first area shows a linear increase of output signal according to the increase of input signal. The compressor in this region is not active. The gain is maximum (see also Figure 33)
2) The second area is characterized by a constant output level and in this region the compressor is active, its output is kept at 200mV$_{PP}$.
The gain is changing from the maximum to the minimum value (see also Figure 33).
3) The third part shows again an increase of the ouput level and in this part the AGC of compressor is no more active. The gain is now minimum (see also Figure 33).

If we apply a modulated signal at the input of the compressor we can see the particular behaviour shown in Figure 35.

As the input signal is increasing the compressor output follows it lineary up to the AGC threshold (1mV$_{RMS}$), then the output is maintained at constant value (200mV$_{PP}$). When the input signal is higher than 11mV$_{RMS}$ the output signal starts to increase. For output signal higher than 250mV$_{PP}$ its distortion increases, but by an external resistor between Pin 16 and GND it is possible to increase this thereshold up to 400mV$_{PP}$ (by 100k). By another external resistor between Pins 16 and 17 it is possible to increase the output level within the range of the dynamic.

Finally the Figure 35 shows the output of Rx compressor with a modulated signal in the input of compressor. This behaviour assures a compensation of the line losses.

Note : The gain is fixed internally

In order to clarify the utility of these compressors we can say that these compressors work with maximum gain when the level of input signal is low and add a variable attenuation (dG), up to reach minimum gain, when the level of the output signal is high.
This value of attenuation is an important parameter for a correct behaviour of hands-free system as you'll see later when we'll speak about attenuators.

V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Figure 30 : Gain of Tx Compressor

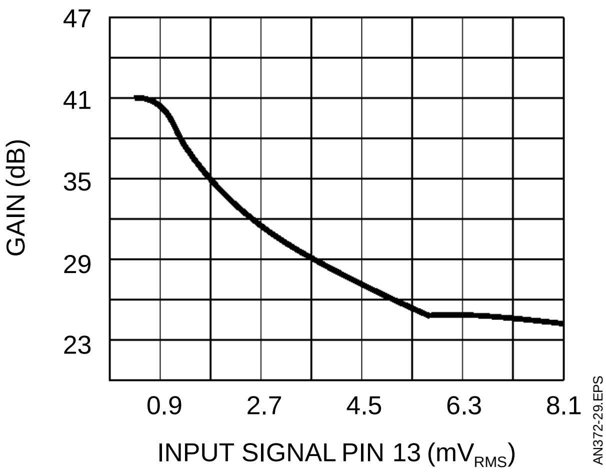


Figure 31 : Dynamic of Tx Compressor

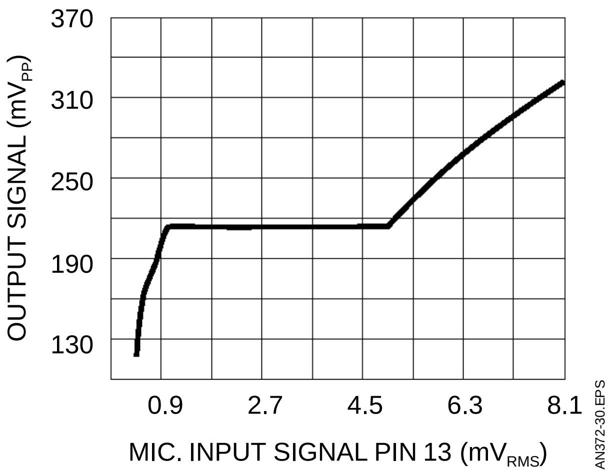
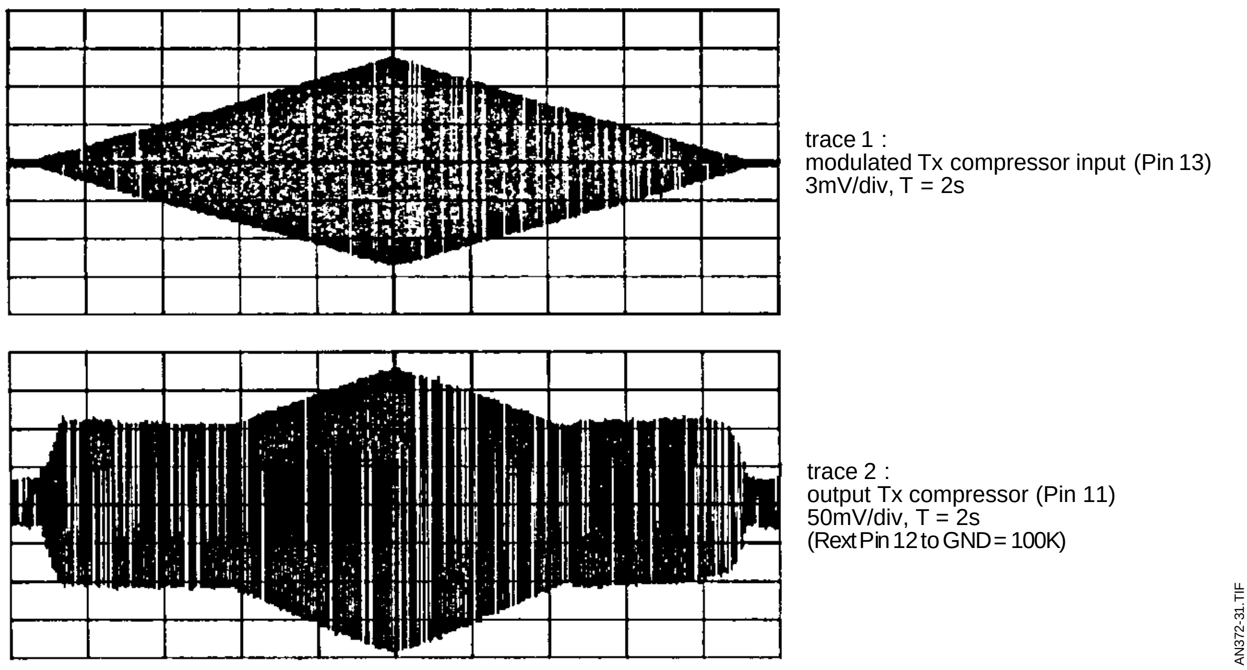


Figure 32 : Dynamic of Tx Compressor



V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Figure 33 : Gain of Rx Compressor

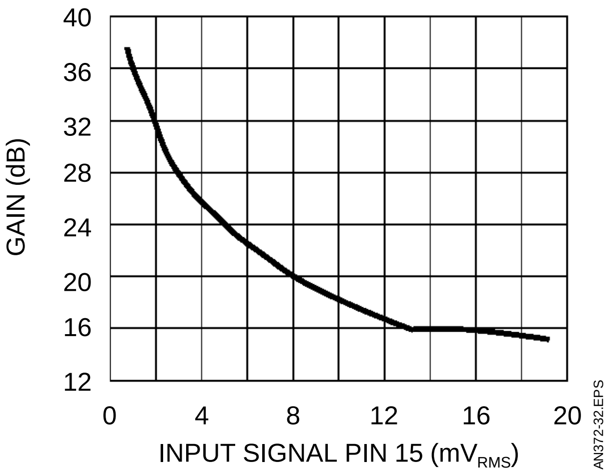


Figure 34 : Dynamic Rx Compressor

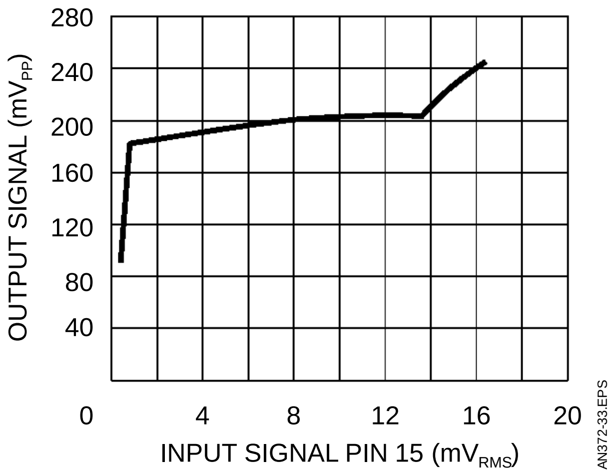
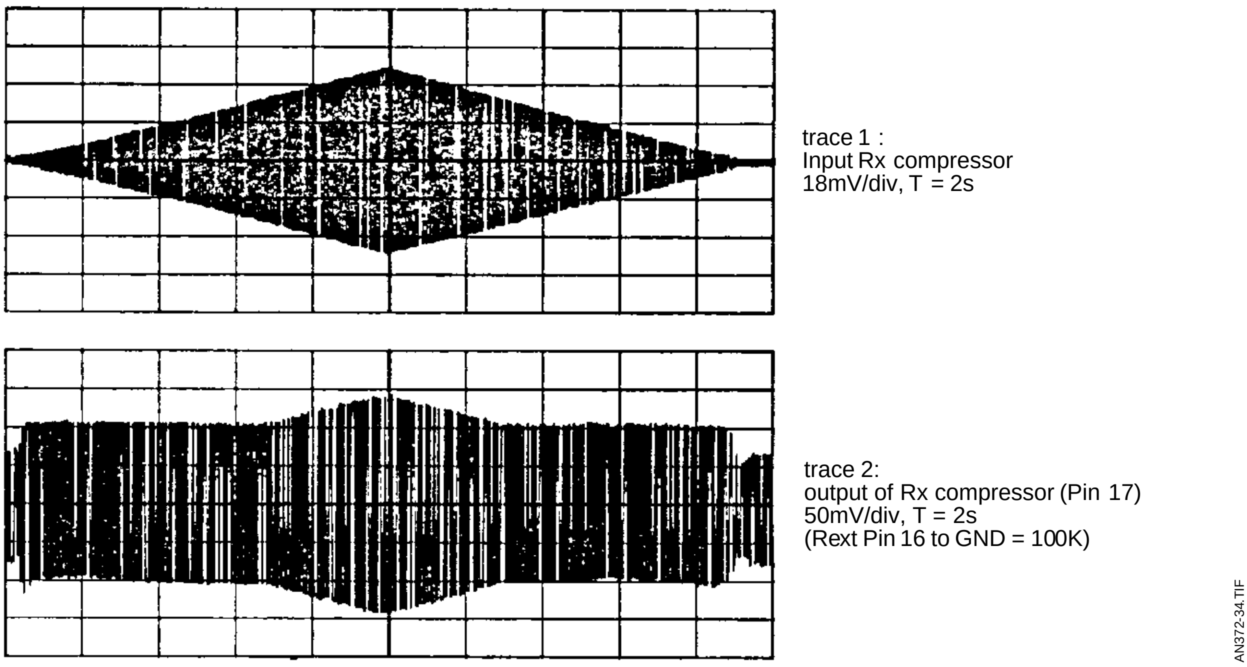


Figure 35 : Dynamic of Rx Compressor



V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

V.1.3 - Tx / Rx Gain in Handset Mode

This system can switch in handset mode by a chip select switch at Pin 2.

Tx path

In this case the measurement points are :
- Pin 21 as input of the microphone in handset mode,
- Pin 3 as output of the transmit path.

The following figure shows the gain of transmit preamplifier in handset mode. In this operating mode the Tx compressor is completely disabled and the microphone input signal is amplified by a constant value.

Note : It's not possible to change externally this value (see Figure 36) .

Rx path

In handset mode the pins of receiving path are the same of the hands-free case, that is, Pins 15 and 26 respectively as input and as output of the signal. In this operating mode the compressor doesn't work like before but its gain is kept at constant value indipendently of the input signal.

This constant value can be changed externally with a resistor connects between Pin 2 and Pin 28.

The Figures 36-37 show three different behaviours of the Rx gain with three different values of the external resistor.

This resistor doesn't affect the Tx gain.

V.2 - Attenuation and Volume Control

V.2.1 - Programmable Attenuation on the paths (AS)

Refering to the block diagram (see also section III.2) both channel have an attenuation block after the compressors, A_T for Tx path and A_R for Rx path. These blocks are controlled by an attenuation control block that manages the correct value of an attenuation to insert in them.

There are three factors that affects this value, that is to say :
- The external resistor (Pin 24) that establishes a particular value of attenuation AS0 when the gains of the compressors are at the maximum value (see also section III.2).
- The value of the attenuation introduced by the comporessors when are not working at maximum gain (see also section III.1).
- The external potentiometer that manages the volume control in Rx path.
- The operating modes (Tx, Rx or Idle) that manage how to share the total attenuation AS between the A_T and A_R blocks.

Taking in account these factors the conditions of the measurement the figure shows are :

Set up condition

a) Gain of the compressors G1 and G2 at the maximum value (attenuation dG1 = dG2 = 0dB).
b) Operating mode is hands-free receiving. In this way the attenuator active is only the transmitting one. In fact like we have said before (see also section III.1) when the system is in Rx mode, A_R = 0dB and vice versa when it's in Tx mode A_T = 0dB.
c) Volume is at maximum value in Figure 38 and it is at its minimum Figure 39 in order to demonstrated its influence on the attenuation.
d) The external resistor, connected between Pin 24 and GND, is equal to 7.4kΩ in Figures 38 - 39 and it is equal zero in Figure 40 in order to insert two different values of attenuations AS0 (see Table 1).

Figure 36 : Gain of Tx Amplifier in Handset Mode

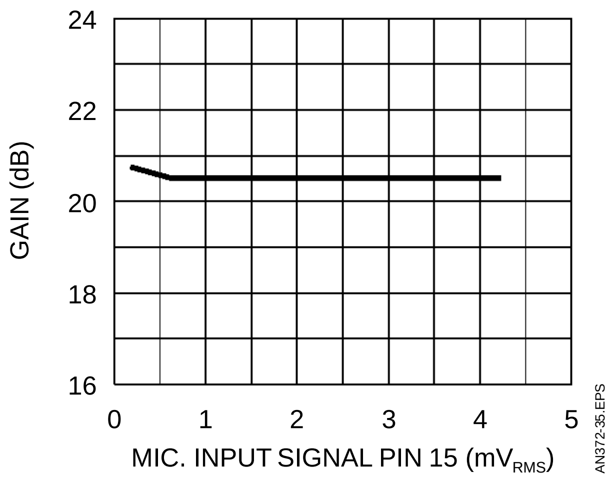
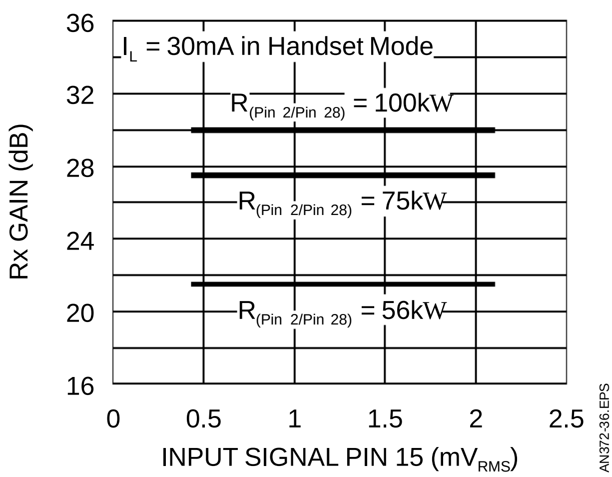


Figure 37 : Gains of Rx Compressor in Handset Mode



V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Rx Path

The Figures 38 and 39 show that when the volume control circuit adds attenuation in the Rx path (vol = min. Figure 39) the attenuation inserted in the Tx path decreases (see traces 2;4).
Note : Rext (Pin 24) = 7.4kΩ ; Volume max.

The Figure 38 and 40 show that when the external resistor programs a lower value of AS0 (Figure 40) the Tx attenuator A_T is reduced and this not affect the Rx attenuation (A_R = 0dB).

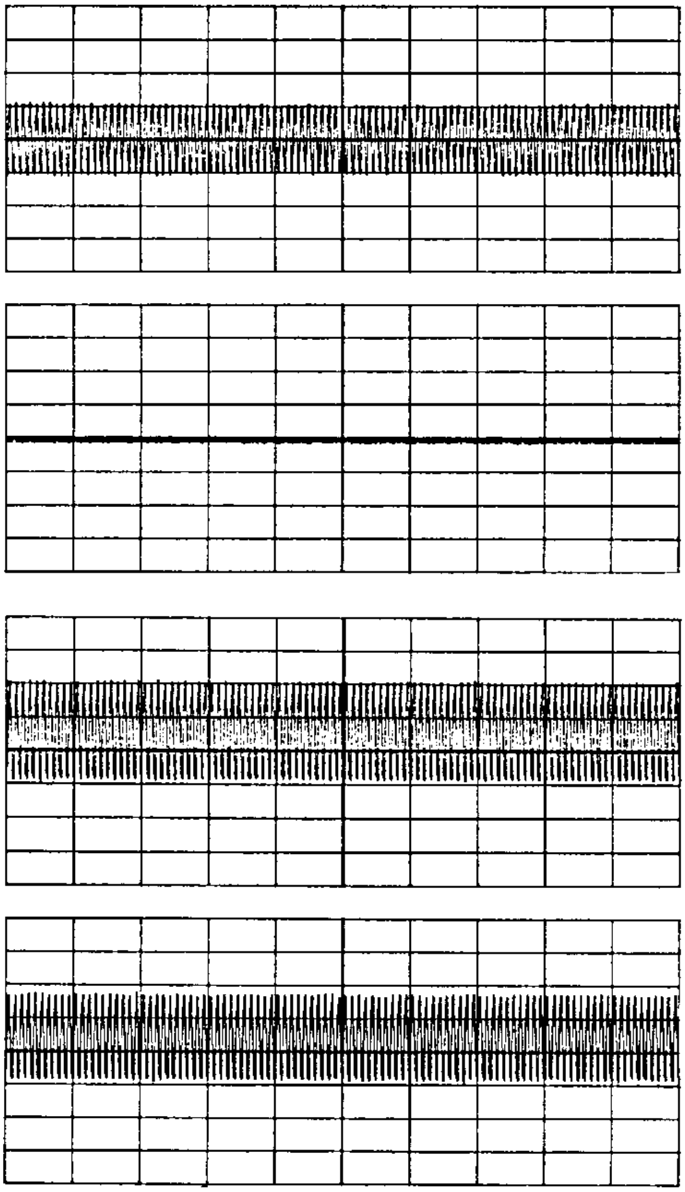
The figures show three different operating states :

- It's possible to note that the input signal of Rx attenuation (Pin 25) is the same of the output one (Pin 26) while the output signal (Pin 3) of Tx attenuator is attenuated (see the table) respect to the input signal (Pin 4).
- The value of the attenuation introduced on Tx path is reduced when RAS, connected from Pin 24 to gnd, is a short circuit (see table).

Note : Rext (Pin 24) = Short Circuit Volume max.

The value of the attenuation introduced on Tx path is again reduced when the volume has its minimum value (potentiometer at Pin 23).

Figure 38



trace 1 :
Input of Tx attenuator (Pin 4)
100mV/div, T = 10ms

trace 2 :
Output of Tx attenuator (Pin 3)
100mV/div, T = 10ms

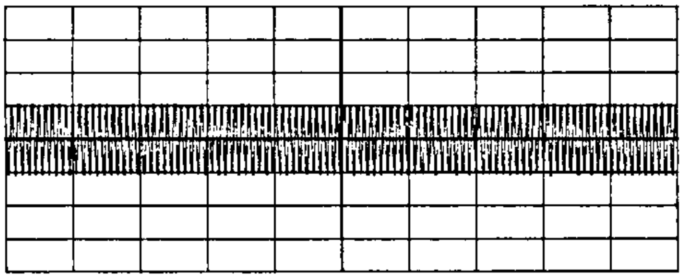
trace 3 :
input of Rx attenuator (Pin 25)
100mV/div, T = 10ms

trace 4 :
output of Rx attenuator (Pin 26)
100mV/div, T = 10ms
V_{OL} max ASO = 37dB

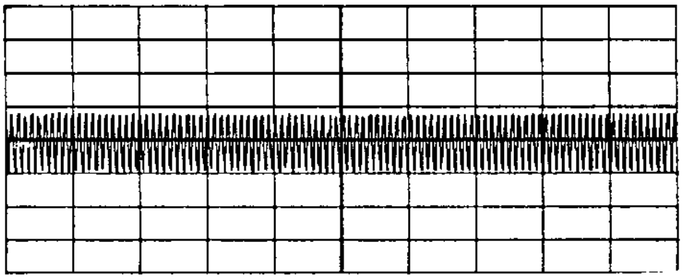
372-37A.TIF / 372-37B.TIF

V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

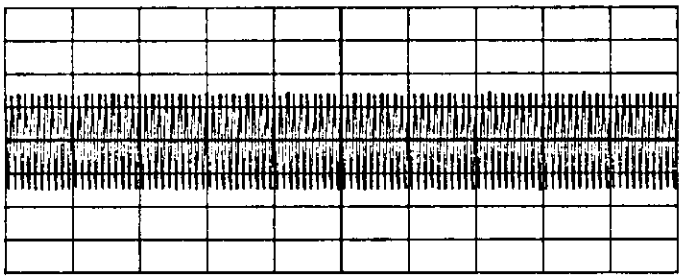
Figure 39



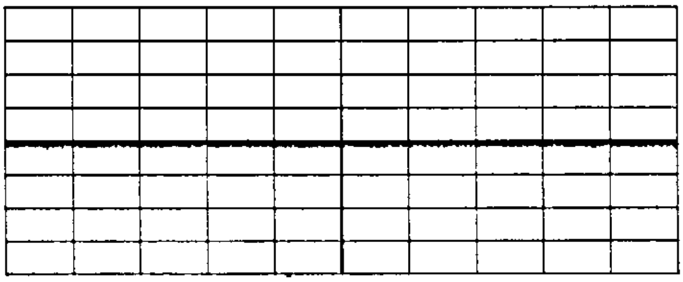
trace 1 :
Input of Tx attenuator (Pin 4)
100mV/div, T = 10ms



trace 2 :
Output of Tx attenuator (Pin 3)
100mV/div, T = 10ms



trace 3 :
Input of Rx attenuator (Pin 25)
100mV/div, T = 10ms



trace 4 :
Output of Rx attenuator (Pin 26)
100mV/div, T = 10ms
(V_{OL} min ASO = 37dB)

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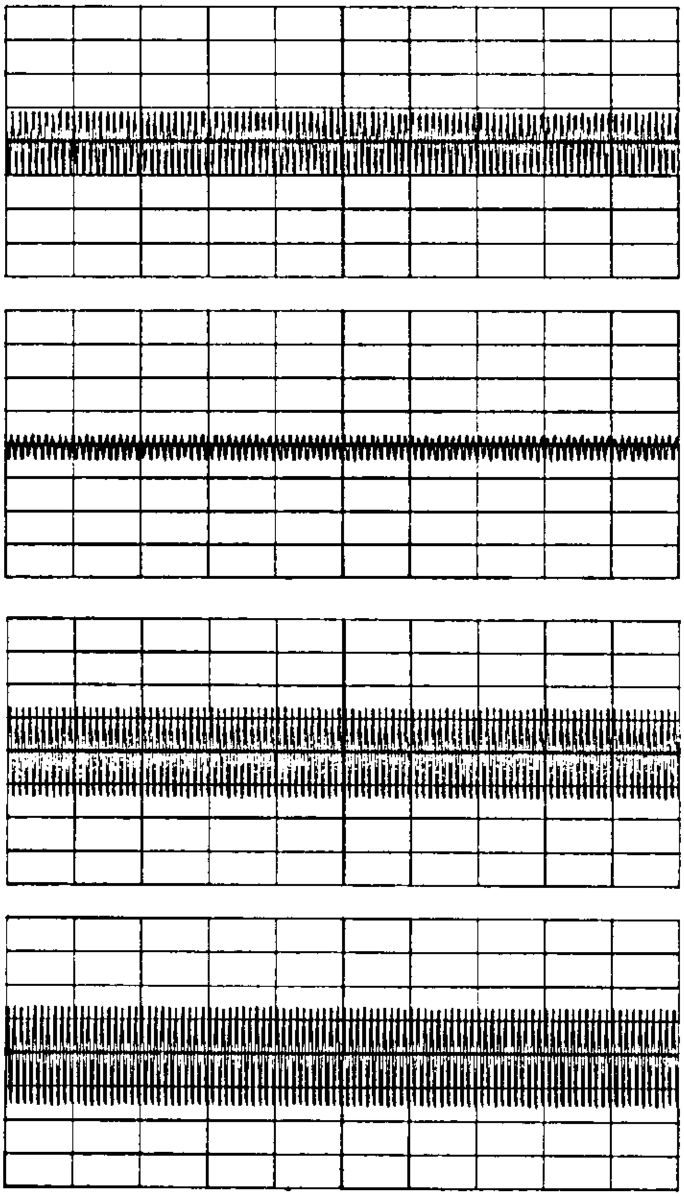
V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Figure 40



trace 1 :
input of Tx attenuator (Pin 4)
100mV/div, T = 10ms

trace 2 :
output of Tx attenuator (Pin 3)
100mV/div, T = 10ms

trace 3 :
input of Rx attenuator (Pin 25)
100mV/div, T = 10ms

trace 4 :
output of Rx attenuator (Pin 26)
100mV/div, T = 10ms
(V_{OL} max ASO = 0dB)

372-39A.TIF / 372-39B.TIF

V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

V.2.2 - Programmable Attenuation of the Tx/Rx Hysteresis Control Blocks (HYT, HYR) (see also section III.4.2)

Refering to the block diagram both channels have a hysteresis block that manages the speech threshold for a correct switch of the system.

The measurement points are :
- Pin 11 as output of Tx compressor
- Pin 17 as output of Rx compressor
- Pin 7 as the timing to observe the correct transition from Tx to Rx and vice versa

There are two factors which influence this behaviour, that is to say :
- Operating mode (Tx or Rx) because in function of the status the correspondent hysteresis block is disabled (see also section III.4.3)
- External resistors at Pin 6 for receiving path and at Pin 5 for transmitting path that establishes the value (HO) to add to default one (- 6dB) (see also section III.4.3)

In fact the final value of the attenuation, that the hysteresis block of the inactive part inserts, depends on these parameters as the general rule shows :

$$HYS = H0 + 6dB$$

The Figure 41 shows how this block acts on the switch of the system. The H0 value was set to 0dB both for Tx and Rx paths. In fact we have the switch when the signals are each other different about 6dB.

Depending on the part we wish to favour, the attenuation inserted (H0#0dB) can be changed by external resistors (at Pin 5 and Pin 6).

V.3 - Rise and Fall Times

V.3.1 - Rise and Fall Times of Tx Compressor (see also section IV.4)

The value of the time constant depends on the value of an external capacitor at Pin 8 that is charged by an internal current source of 1.25µA (t_R) and discharged by another internal current source of 100µA (t_F)).
The ratio is equal to 80 and so : $80\ t_F = t_R$ during t_R the compressor increases its gain and viceversa during t_F the compressor decreases its gain in order to maintain a constant output voltage level.

The Figure shows a slow rise time and a very fast fall time, corresponding to the peak of the output compressor signal trace. From an application point of view this timing avoids that the compressor increases its gain quickly when there is only environment noise during pauses within spoken words.

In fact when at the input there is a very low signal the compressor works with its maximum gain as we have said before, such a choice of the time constants avoids an unpleasant fast amplification of the noise when nobody is speaking.

The external capacitor can increase or decrease the value of the time constants but not the ratio between them.

The pins of the measurements are respectively :
- Pin 11 as output of compressor of transmit path
- Pin 8 as pin of time constants of Tx compressor
- Pin 13 as input of compressor of transmit path

The Figure 42 shows the rise time and fall time of the compressor in transmit path.

No signals are connected to the receive input during the measurements.

V.3.2 - Rise and Fall Time of Rx Compressor

In the same way of the previous analysis it's possible to investigate the constant time of the Rx compressor.

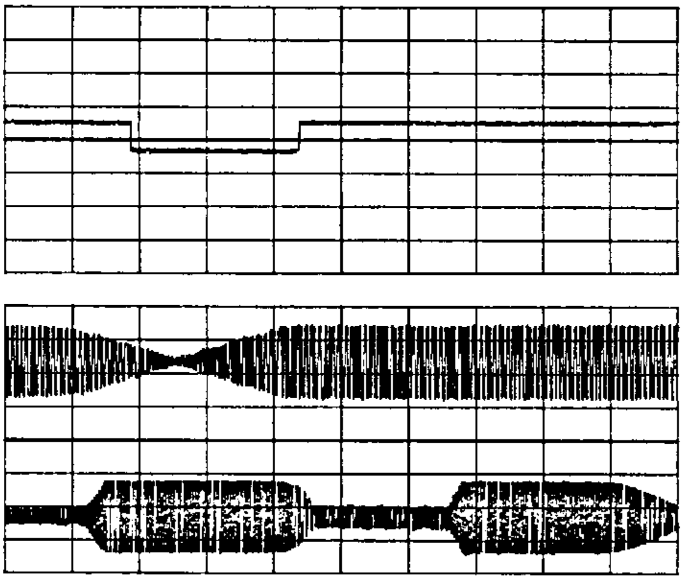
The philosophy of the design is the same, but the external capacitor is connected to Pin 20. The ratio between rise time and fall time is 80 independently of the value of the capacitor.

The Figures 43 and 44 enphasized the fall time and the rise time with their ratio. In fact it is shown an expanded time scale to note as well as possible the fast fall time (Figure 43).

The pins of the measurements are respectively :
- Pin 17 as output of the compressor of receive path
- Pin 18 as output of the peak detector of receive path

V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Figure 41



trace 1 :
Timing pin (Pin 7)
500mV/div, T = 500ms

trace 2 :
output of Tx comparator (Pin 11)
100mV/div T: 500ms

trace 3 :output of Rx comparator (Pin 17)
100mV/div, T = 500ms

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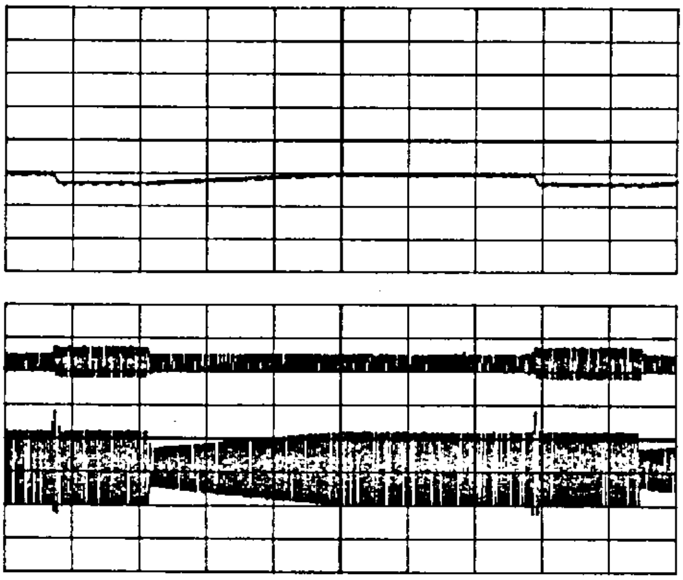
Figure 42



trace 1 :
Time constants of Tx comparator (Pin 8)

trace 2 :
Input of Tx comparator (Pin 13)
5mV/div, T = 200ms

trace 3 :
Output of Tx comparator (Pin 11)
100mV/div, T = 200ms

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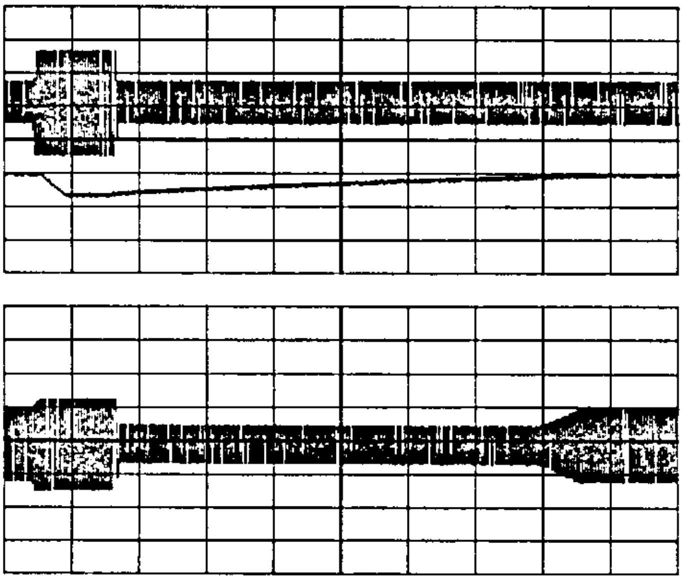
V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Figure 43



trace 1 :
Input of Rx comparator (Pin 15)
16mV/div, T = 100ms

trace 2 :
Time constants of Rx comparator (Pin 20)
200mV/div, T = 100ms

trace 3 :
Output of Rx comparator (Pin 17)
100mV/div, T = 100ms

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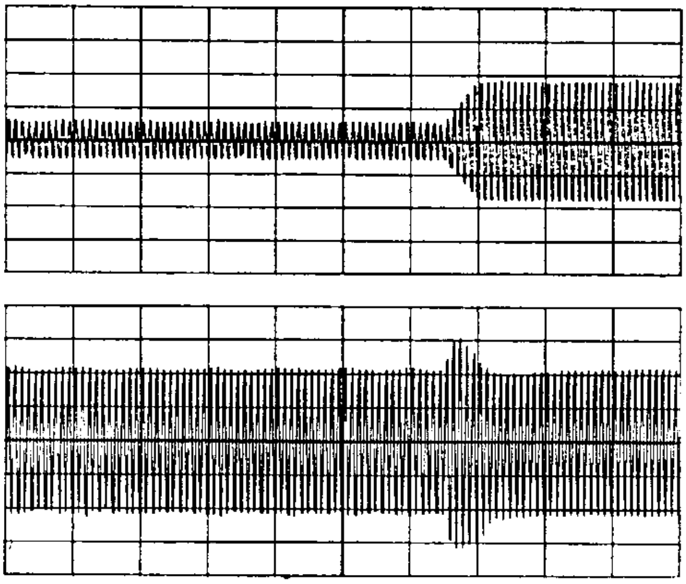
Figure 44



trace 1 :
Input of Rx comparator (Pin 15)
8mV/div, T = 10ms

trace 2 :
Output of Rx comparator (Pin 17)
50mV/div, T = 10ms

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V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

V.3.3 - Rise and Fall Time of Tx Peak Detector

The measurement points are respectively :
- Pin 10 as output of Tx peak detector,
- Pin 11 as output of Tx compressor.

The constant times are fixed by an external RC network at Pin 10 as below :

$$t_A = \frac{CV}{I},\ t_D = 2\ RC$$

V = 100mVdc fixed internally,
I = 20µA fixed internally, C = 470nF, R = 75kΩ.

The input signal has been modulated to obtain a fast increase of the output signal of the compressor as the Figures 45-46 show. The Figure 46 enphasized, with an expanded time scale, the rise time.

V.3.4 - Rise and Fall Time of Rx Peak Detector

In the same way of the previous analysis the rise time and the fall time of Rx peak detector are shown in the Figure 47.

The measurement points are respectively :
- Pin 18 as output of Rx peak detector,
- Pin 17 as output of Rx compressor.

The constant times are fixed by an external RC network at Pin 18 as the rules above.

Also in this case the input signal has been modulated to obtain a fast increase of the output signal of compressor as the figure shows.

V.3.5 - Delay Time between Tx/Rx and Idle Mode Transitions

The three general modes of the system (transmit, receive, idle) are managed depending on the information from three comparators that is : (see also the block diagram)
- Transmit/receive comparator,
- Speech/Noise comparator on the Tx path,
- Speech/Noise comparator on the Rx path.

Depending on the outputs of these blocks the attenuation block manages the channels attenuations.

The constant times are fixed by an external RC network at Pin 7 as the rule below.

$$t_{RISE} = t_R = \frac{CV}{I} \quad \text{and} \quad t_{FALL} = t_F = 2.2\ RC$$

V = 220mVdc fixed internally,
I = 50µA fixed internally, C = 1µF, R = 430kΩ.

The Figure 48 shows the transition between Idle to Tx modes and the Figure 49 from Rx to Idle.

V.3.6 - Delay Time between Tx and Rx Mode Transitions

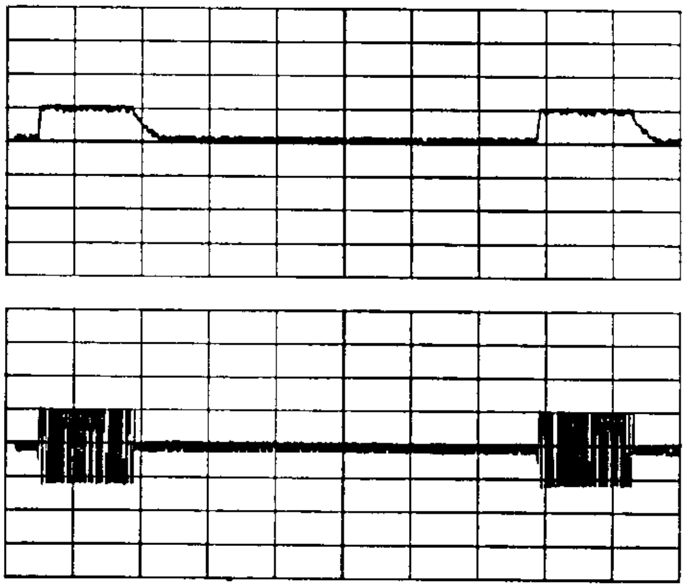
The Figure 50 reasumes the behaviour of the transitions of the system

The measurements points are :
- Pin 7 timing pin
- Pin 17 output of Rx compressor
- Pin 11 output of Tx compressor

The Tx and Rx input signals was choosed in order to show the Rx to Tx, Tx to Idle and Idle to Rx transitions.

V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Figure 45



trace 1 :
Output of Tx peak detector (Pin 10)

trace 2 :
Output of Tx comparator (Pin 11)

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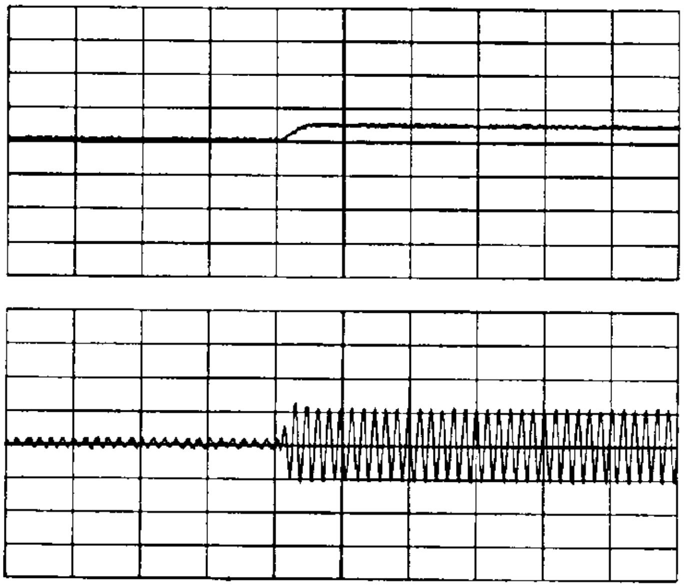
Figure 46



trace 3 :
Output of Tx peak detector (Pin 10)
200mV/div, T = 10ms

trace 4 :
Output of Tx comparator (Pin 11)
100mV/div, T = 10ms

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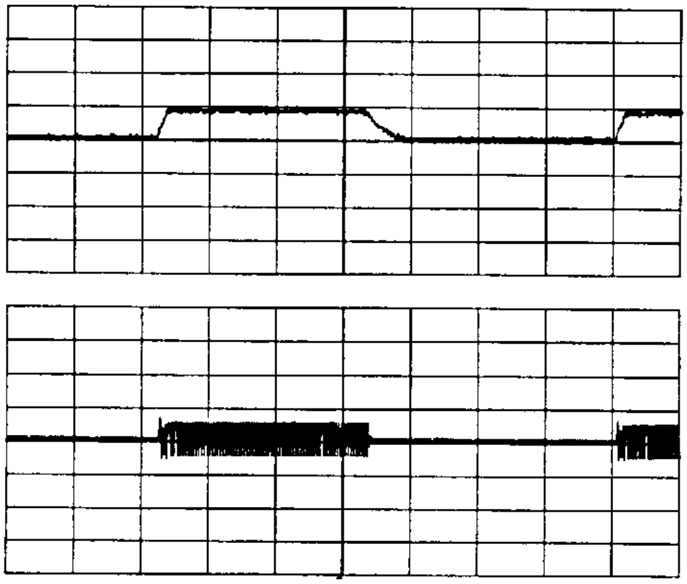
V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Figure 47



trace 1 :
Output Rx peak detector (Pin 18)
100mV/div, T = 200ms

trace 2 :
Output of Rx comparator (Pin 17)
200mV/div, T = 200ms

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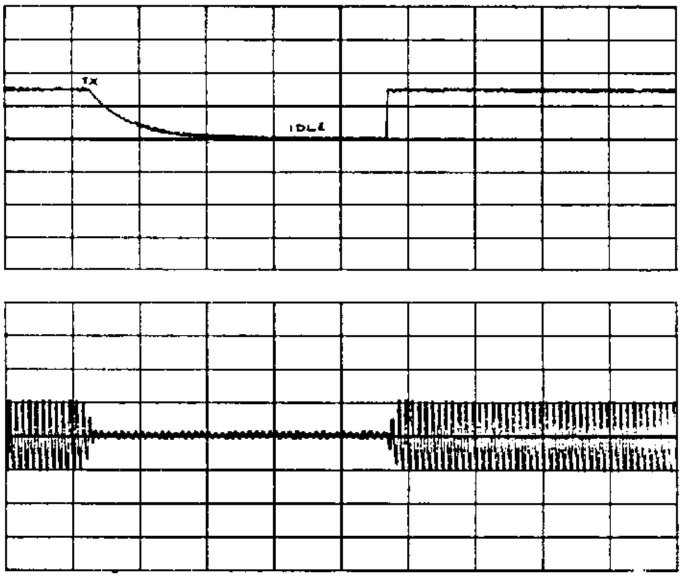
Figure 48



trace 1 :
Timing pin. (pin 7)
Tx - IDLE transition
200mV/div, T = 500ms

trace 2 :
Output of Tx comparator (Pin 11)
100mV/div, T = 500ms

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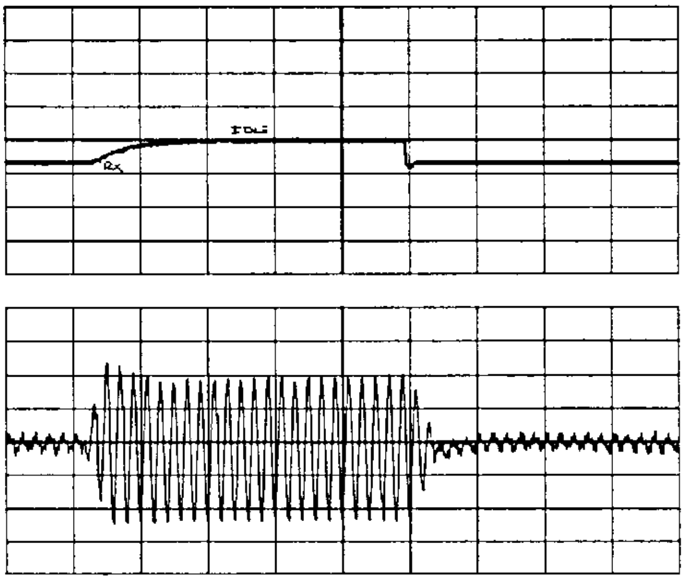
V - INFLUENCE OF THE EXTERNAL COMPONENTS (continued)

Figure 49



trace 2 :
Timing pin (Pin 7)
Rx - Idle

trace 1 :
Output of Tx comparator (Pin 18)
50mV/div, T = 10ms

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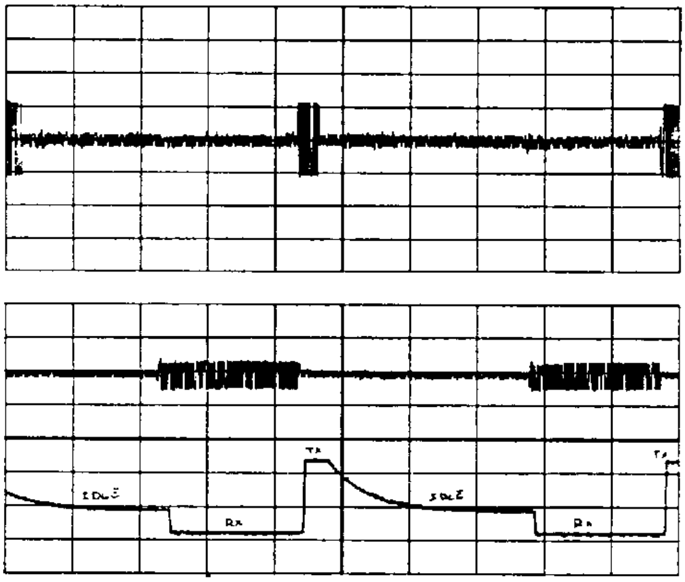
Figure 50



trace 1 :
Output Tx compressor (Pin 11)

trace 2 :
Output of Rx comparator (Pin 17)
10mV/div, T: 50ms

trace 3 :
Timing pin (Pin 7)

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VI - APPLICATIONS

VI.1- Hands-free Particular Effects

VI.1.1 - The Larsen Effect

This effect occurs in applications when the acoustic and sidetone coupling effects are enough to produce a loop gain greater than unity, resulting in a ”howling effect” in the loudspeaker. Usually the acoustic coupling from the loudspeaker to the microphone is the principal cause of this instability.

To avoid this condition the value of the resistance connected to Pin 24 has to be increased until the instability disappears, due to the increase of the AS0 value.

VI.1.2 - ”Chopped Speech” Effect

During a two-way conversation two effects may be encountered :

- chopping of the received signal,
- chopping of the transmitted signal.

This effect is produced by the loss of parts of a spoken word due to transitions from one active mode to the other (transmit to receive, or vice-versa), caused by incorrect contol of the signals level in input of the comparators, bearing in mind the hysteresis effect which controls the speech level in the not active channel (see section III.4).
When, in the receive mode, the received signal is chopped the effect is due to insufficient hysteresis in the transmit channel. The level of any speech signal present in the transmit signal is not sufficiently reduced to maintain the dominance of Rx channel. In this case the level of the programmable attenuation H_{OT} must be raised, by decreasing the value of the resistance connected to Pin 5 (see section III.4.3).
In a similar way when the system is in transmit mode, if the chopping effect occurs in the transmit channel the level of the programmable attenuation, HOR has to be raised in the receive channel, by increasing the value of the resistor connected to Pin 6 (see section III.4.3). take note an excessive value of hysteresis can hold up the switching.

VI.1.3 - ”Walky-Talky” Effect

Under certain conditions the switching of attenuation between the transmit and receive channels may produce an unpleasant effect. A sudden apparently total loss of reception each time speech beginning, produces a half-duplex effect similar to that obtained with Walky-Talky sets. The basic cause of this effect is an excessive value of attenuation switched from one channel to the other when the active mode switches from receive to transmit.
To optimize any particular implementation based on TEA7540 chip set the following adjutsments should be made :

- The attenuation AS0 should first be adjusted to have the minimum value necessary to respect the loop gain stability requirement (see section VI.1.1). AS0 should then be increased by up to a further 3dB in order to reduce the distortion produced by the system close to the Larsen Effect limiting condition.
- The average speech level at the compressor input should then be adjusted to obtain the half value of compressing range.

To carry out this adjustment connect an oscilloscope probe to both the compressor gain control inputs, CCE and CCR (Pins 8 and 20 respectively).

Adjust the level of the speech signal inputs to the compressors such that, for a typical average speech level, an average level of V_{REF} -110mV is obtained on Pin 8 and V_{REF} -50mV is obatained on Pin 20 of TEA7540.

VI.1.4 - Volume Control Influence

In the receive mode any additional attenuation added (or sutracted) in the Rx attenuator is compensated by an equal attenuation subtracted (or added) in Tx attenuator :
AT = AS0 - dG1 - dG2 - AVOL (see section III.1)

In the transmit mode, however, the attenuation AT remains fixed at 0dB and the attenuation AR is still:
AR = AS0 - dG1 - dG2

In resume, the volume control attenuation improve the duplex effect only in the receive mode.

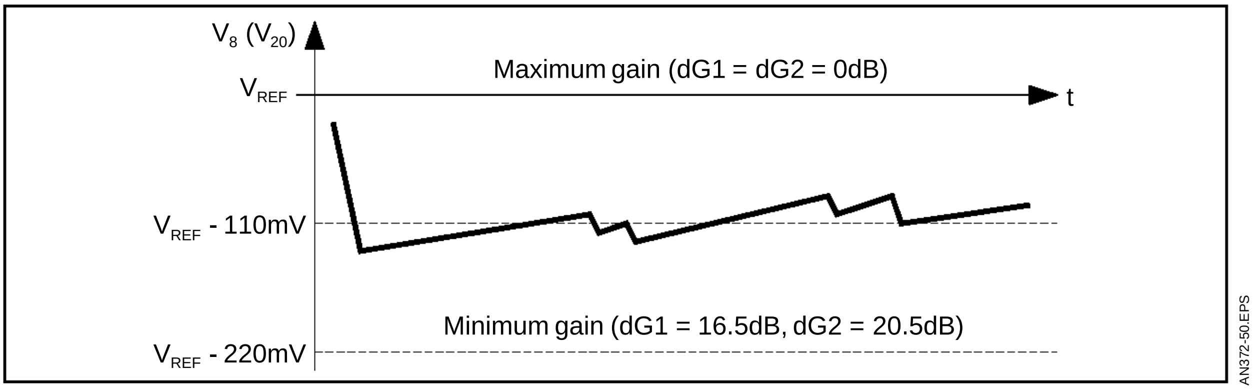
VI.1.5 - Examples

a) Example 1:
In the following example the attenuation AS0 is 45dB. After optimization, the total attenuation given by the attenuators is :
AS = AS0 - dG1max/2 - dG2max/2
hence: AS = 45dB - 8.5dB - 10dB = 26.5dB
In this case, the speech signal of the inactive path is only attenuated and not lost for the listener. Take note, AS is decreasing while the compressor input signals is increasing. This is due to dG1 or dG2 increase depending on the operating mode of the system. The attenuation minimum value of AS is :
AS = AS0 - dG1max/2 - dG2max/2
hence: AS = 45dB - 17dB - 20dB = 8dB

b) Example 2:
In the following example the attenuation AS0 is 37dB. In this case the attenuators give:
AS = 37dB - 8.5dB - 10dB = 18.5dB
and for the speech signal maximum amplitude:
AS = 37dB - 17dB - 20dB = 0dB
Therefore, the presence of this ”quasi full duplex” operation provided that the application is carefully done. This is due to the presence of the compressor, the steady level of transmit signal for a distance variation between speaker and microphone from 0.5meters up to 3meters.

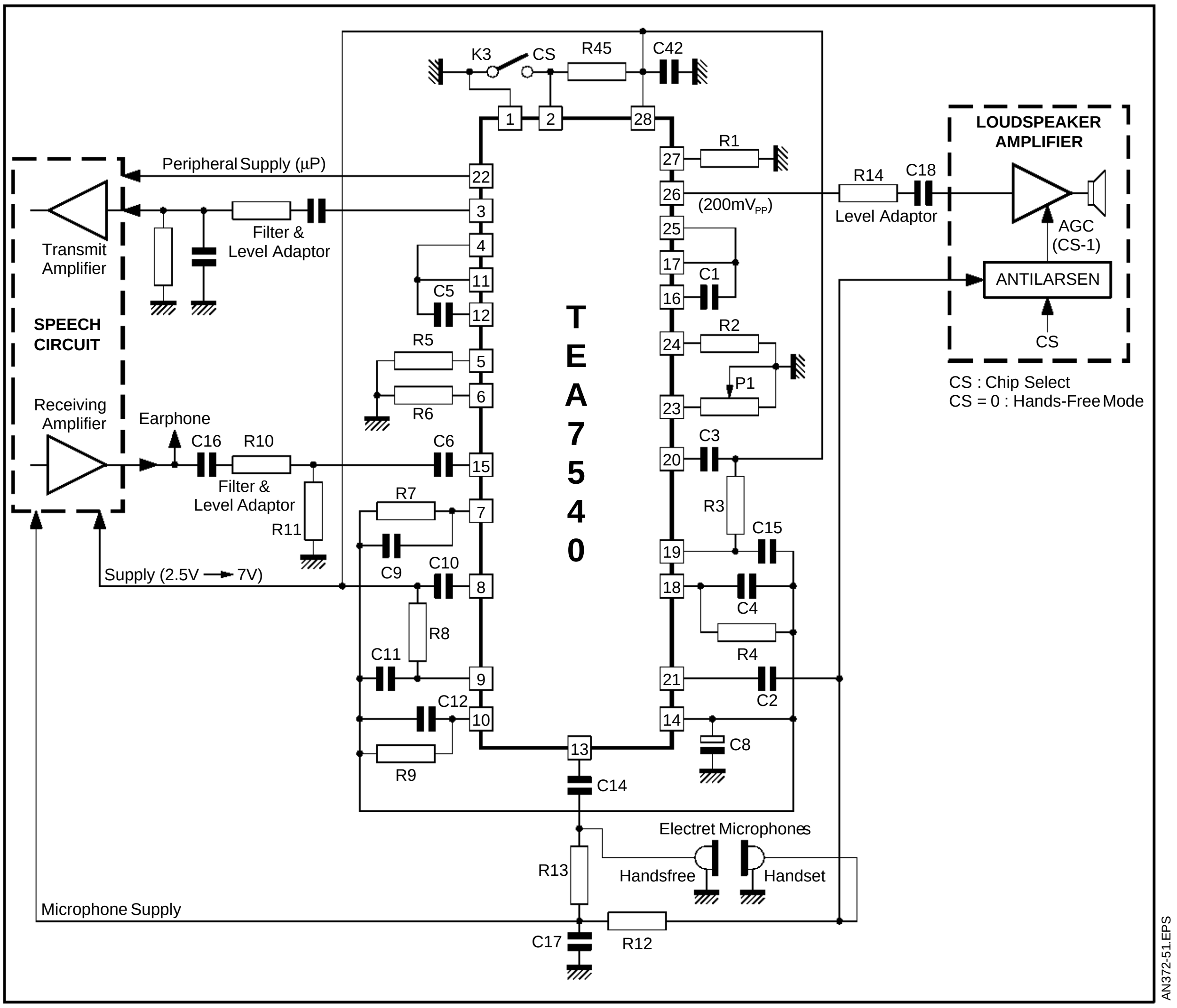
VI - APPLICATIONS (continued)

Figure 51 : Adjustment of comparator input levels



VI.2 - Basic Hands-free Telephone Set

Figure 52 : Basic Hands-Free Schematic



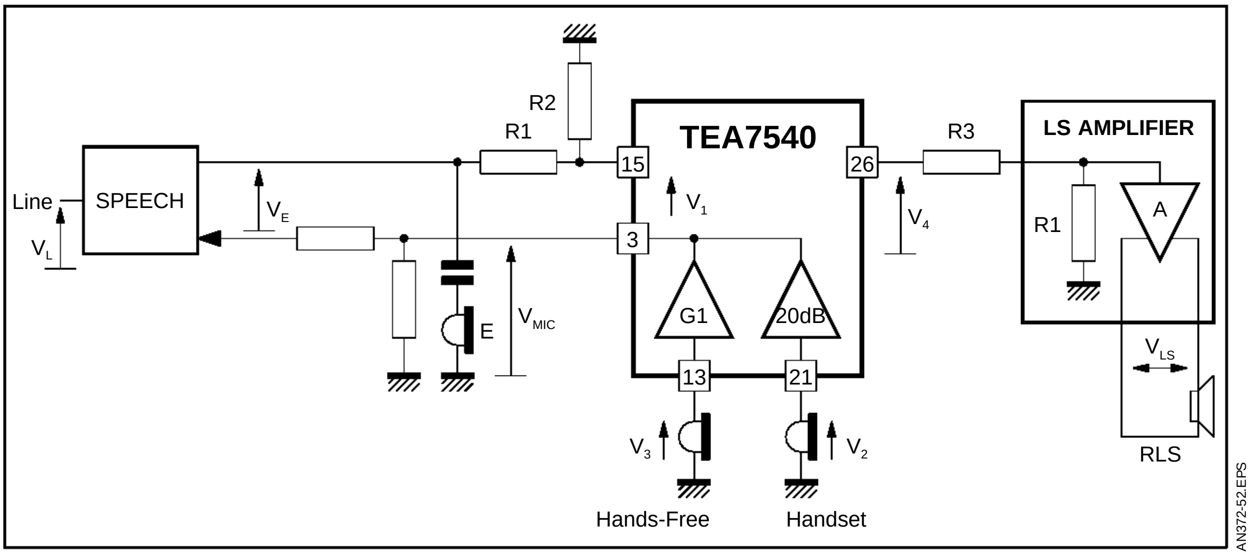
VI - APPLICATIONS (continued)

Table ?? : Component Value (see Figure 52)

Symbol	Value	Description
C1	100nF	coupling and high pass filter
C2	47nF	coupling and high pass filter
C3	10µF	Rx compressor gain rise and decay time
C4	2.2µF	Rx peak detector rise and decay time
C5	100nF	coupling and high pass filter
C6	33nF	coupling and high pass filter
C7	100µF	power supply decoupling
C8	100µF	reference voltage decoupling
C9	470nF	switching time timer (t1, t2, t3)
C10	10µF	Tx compressor gain rise and decay time
C11	68µF	Tx background noise rise and decay time
C12	470nF	Tx peak detector rise and decay time
C13	100µF	microphone supply decoupling
C14	18nF	coupling and high pass filter
C15	68µF	Rx background noise rise and decay time
P1	10K	Rx channel volume control
R1	3.57K	1%, internal bias current source adjusting
R2	9.1K	AS0 attenuation
R3	4.7M	Rx background noise rise time
R4	30K	Rx peak detector decay time
R5	5.6K	Tx hysteresis (attenuation)
R6	1M	Tx hysteresis (attenuation)
R7	820K	Switching time (from speech to idle) timer
R8	4.7M	Tx background noise rise time
R9	75K	Tx peak detector decay time
R10	2.2K	Tx input level adjusting, Hands-free mode
R11	2.2K	Tx input level adjusting, Hands-free mode
R12	56K	Rx pre-amplifier gain, monitoring mode

VI.3 - Monitoring Mode and Hands-Free Gains Compability

Figure 53 : Interface Components Calculation



VI - APPLICATIONS (continued)

a) Tx gain in hands-free mode

The transmit compressor has been designed to have a steady Tx signal level for a distance variation between speaker and microphone from 0.5meter up to 3meters with :

$$25\text{dB} \leq G1 \leq 41.5\text{dB}$$

For this gain range the output signal of compressor is stabilized at the Vmic = 200mVPP (see section IV.1.2).

Hence, the input microphone typical signal range can be calculated easily:

$$1.7\text{mV}_{PP} \leq V_3 \leq 11.2\text{mV}_{PP}$$

b) Tx gain in handset mode

The handset microphone has to provide a voltage, $V_2 = V_{MIC}/20\text{dB}$, in normal acoustic levels conditions to have the same Tx level on line than in hands-free mode.

Therefore, the speech stage should be adapted to the compressor output voltage :

$$V_{MIC} = 200\text{mV}_{PP}$$

Hence : $V_2 = 20\text{mV}_{PP}$

c) Rx gain

R1 and R2 evaluation: the receive compressor has been designed to have a steady Rx signal level such as $V_4 = 200\text{mV}_{PP}$: The typical range of the Rx compressor is :

$$16\text{dB} \leq G2 \leq 36.5\text{dB}$$

We can deduce that the input signal range is :

$$3\text{mV}_{PP} \leq V_1 \leq 32\text{mV}_{PP}$$

Let us take one example: for FRANCE the mean speech line level is equal to about -15dBm (390mV_{PP}). To optimize an application based on TEA7540 chip set, the average speech level at compressor input must be adjusted to obtain the middle value of the compressing range (see section V.1.3).

That is to say : $V_1 = 17.5\ \text{mV}_{PP}$

The Rx speech gain is X dB (according to the country) :

$$20\log_{10}(V_E/V_L) = X\text{dB}$$

For FRANCE XdB = -2dB

Hence $V_E = 300\text{mV}_{PP}$

Hence

$V_E/V_1 = R_2/(R_2+R_1) = 300\text{mV}_{PP}/17.5\text{mV}_{PP} = 17$

We can choose: $R_1 = 16\text{k}\Omega$ and $R_2 = 1\text{k}\Omega$

For FRANCE the MF signal specification is :
$V_L = 84\text{mV}_{PP}$ min.

This value must be included in the compressing range. checking :

$$V_L = 84\text{mV}_{PP} \Rightarrow V_E = 66.7\text{mV}_{PP}$$

hence: $V_1 = V_E/17 = 3.9\text{mV}_{PP}$

In practice, this signal can be greater. Our example permits :

$$V_1 = 32\text{mV}_{PP} \Rightarrow V_L = 800\text{mV}_{PP}$$

d) Available power for loudspeaker and volume control

R3 evaluation : for a standard loudspeaker an average nominal power PLS = 10mW.

Therefore, for each loudspeaker impedance there is an equivalent voltage :

R_{LS}	V_{LS}
25Ω	$1.4V_{PP}$
50Ω	$2.0V_{PP}$
100Ω	$2.8V_{PP}$
150Ω	$3.5V_{PP}$

The SGS THOMSON loudspeaker amplifier, TEA7532, can provide on 50Ω of load 25mW maximum with 3V power supply and 100mW with 5V power supply.

Now, it is easy to calculate R3 in order to obtain V_{LS} from $V_4 = 200\text{mV}_{PP}$:

$$R_3 = R_I \cdot [A \cdot 0.2V_{PP}/V_{LS} - 1]$$

with $R_I = 1\text{k}\Omega$, $12\text{dB} \leq A \leq 32\text{dB}$ programmable

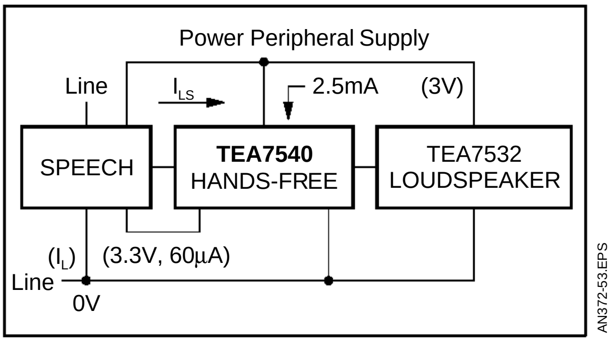
For example with A = 32dB, $V_{LS} = 3V_{PP}$ and 50Ω as load, we must put :

$$R_3 = 1\text{k}\Omega\ [32\text{dB} \cdot 0.2V_{PP}/2] = 4\text{k}\Omega$$

(see French application circuit)

Note : It is possible to increase the VL_S value up to $8V_{PP}$ on 150Ω of load (54mΩ) using $R_3 = 0\Omega$ (see Italian application circuit)

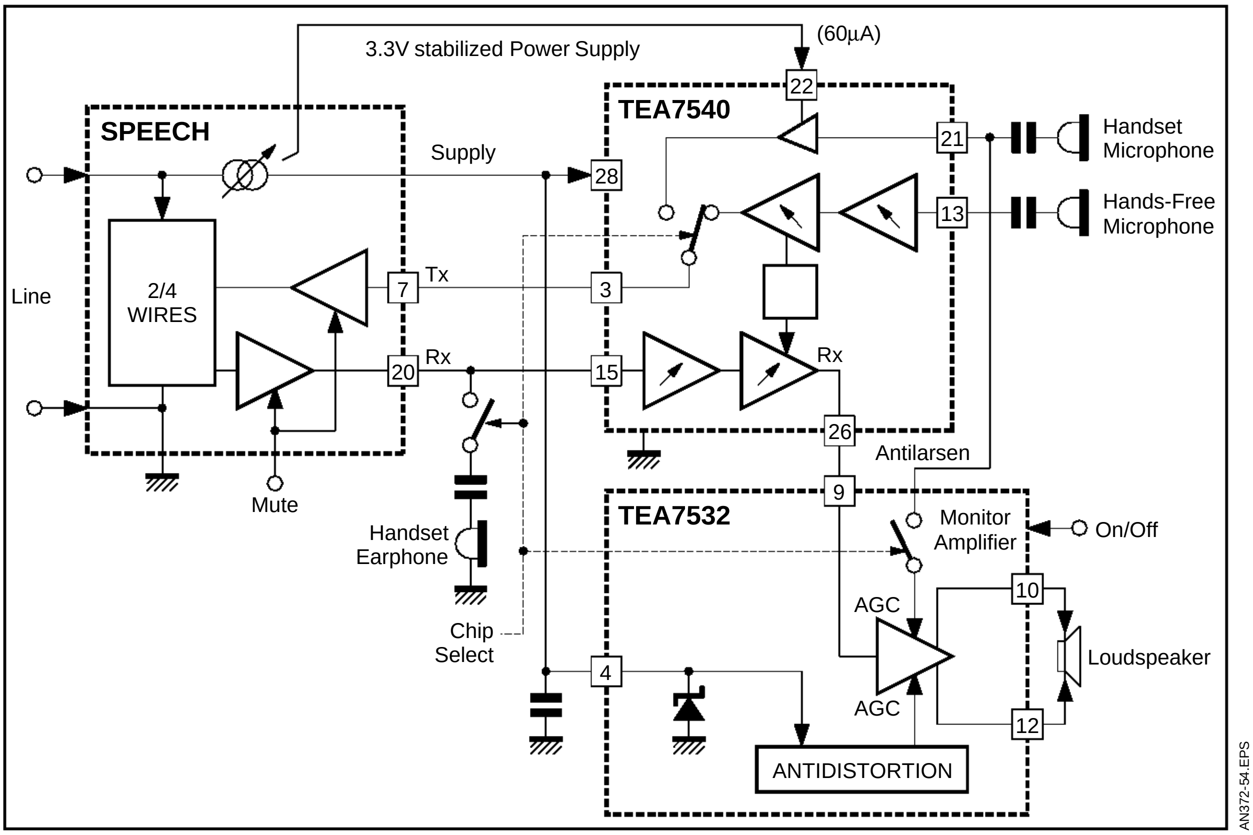
Figure 54 : Low Cost Design Solution



VI - APPLICATIONS (continued)

VI.4 - Hands-Free Option

Figure 55 : SGS-THOMSON Universal Modular Hands-Free Kit (simplified schematic)



TEA7532 is a 16 pins package which provides :

- Loudspeaker amplifier : Antidistorsion system
- Shunt regulator : It may be supplied by a current source. The IC has a zener characteristic which can be used as voltage supply for TEA7540.
- Antiacoustic feed-back system (anti-larsen system). This function is active only in monitoring mode with TEA7540.