UE910 V2, CE910 Audio Settings Application Note

80419NT11272A Rev. 0 – 2014-07-24

Making machines talk.

APPLICABILITY TABLE

PRODUCTS
UE910-EU V2
UE910-NA V2
UE910-EU V2 AUTO
CE910-DUAL
CE910-SC

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Contents

1. Introduction

1.1. Scope

Scope of this document is to provide an overview of the technical aspects to consider while implementing audio functionalities onto applications integrating Telit modules specified in the aforementioned Applicability Table.

1.2. Audience

This document is intended for Telit customers who are about to implement audio functionalities into their projects.

1.3. Contact Information, Support

For general contact, technical support, to report documentation errors and to order manuals, contact Telit's Technical Support Center (TTSC) at:

TS-EMEA@telit.com
TS-NORTHAMERICA@telit.com
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Alternatively, use:

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For detailed information about where you can buy the Telit modules or for recommendations on accessories and components visit:

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Our aim is to make this guide as helpful as possible. Keep us informed of your comments and suggestions for improvements.

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1.4. Document Organization

This document contains the following chapters:

Chapter 1: "Introduction" provides a scope for this document, target audience, contact and support information, and text conventions.

Chapter 2: "The Audio Interface" gives a functional overview of the Audio Interface.

Chapter 3: "Why HS, MT and HF" tries to give an explanation from an historical point of view about the fundamental differences between these acronyms.

Chapter 4: "Electrical Characteristics" describes the equipment providing considerations about the input and output lines of the equipment.

Chapter 5: "Single Ended or Differential" provides a discussion about the use of these configurations.

Chapter 6: "Microphone Amplifier" provides a discussion about the use of the available configurations when designing microphone amplifiers.

Chapter 7: "Speaker amplifier" provides a discussion about the use of the available configurations when designing speaker amplifiers.

Chapter 8: "Electret Microphone" provides a technical description and considerations about electrets microphones (ECM).

Chapter 9: "Input Paths Guide Lines" indicates the Differential one as a better configuration choice and provides practical suggestions.

Chapter 10: "Output paths guidelines" indicates the Differential one as a better configuration choice and provides practical suggestions.

Chapter 11: "Echo cancellation" explains how to overcome the acoustic echo signal disturbance implementing an Acoustic Echo Controller in the firmware of the baseband chip.

Chapter 12: "Safety Reccomendations"

1.5. Text Conventions

Danger – This information MUST be followed or catastrophic equipment failure or bodily injury may occur.

Caution or Warning – Alerts the user to important points about integrating the module, if these points are not followed, the module and end user equipment may fail or malfunction.

Tip or Information – Provides advice and suggestions that may be useful when integrating the module.

All dates are in ISO 8601 format, i.e. YYYY-MM-DD.

1.6. Related Documents

- UE910 V2 series Hardware User Guide, **1vv0301072 and 1vv0301065**
- CE910 series Hardware User Guide, **1vv0301010.**
- AT Commands Reference Guide, **80419ST10124a and 80399ST1011a.**
- Telit EVK2 User Guide, **1vv0300704**

1.7. Document History

Revision	Date	Changes
r0	24 July 2014	Initial release

2. The Audio Interface

2.1. Introduction

The Audio Interface is one part of the Baseband System. It comprises the digital and analog circuits for receive and transmit audio operation and ringing.
In this document only the analog blocks will be dealt with, but at its end you can find some paragraphs that describe the functionalities and suggest the implementations of the Echo Canceller/Noise Suppressor modules that allow the use of the Telit modules in Handsfree or Car kit environments, even if they are digital parts of the firmware section.
In the rest of the document we shall refer to *Baseband Audio Interface*.

2.2. Functional Overview

The ***Audio Interface*** can be considered organized in three sections:

- Interface to processor cores
- Digital filters
- Analog circuits

As declared, in this document we shall describe only the analog circuits that transform the signals to and from the digital part of Baseband chip into analog signals usable by both receive (*Downlink*) and transmit (*Uplink*) audio sections.
The following main functional blocks could be recognized:

- Microphone supply generation
- Analog Input filtering and buffering
- Analog Output filtering and buffering
- Ringer generation

The ***Audio Interface*** has two major operation modes, featuring a high quality DAC/ADC conversion with buffering stages, allowing the connection of external acoustic transducers as shown in figure 1:

- *Power-down mode* . All analog parts are powered down and all digital and analog part clocks are switched off.
- *Audio mode* . All digital filters are connected to the interface buffers and the analog part is enabled.

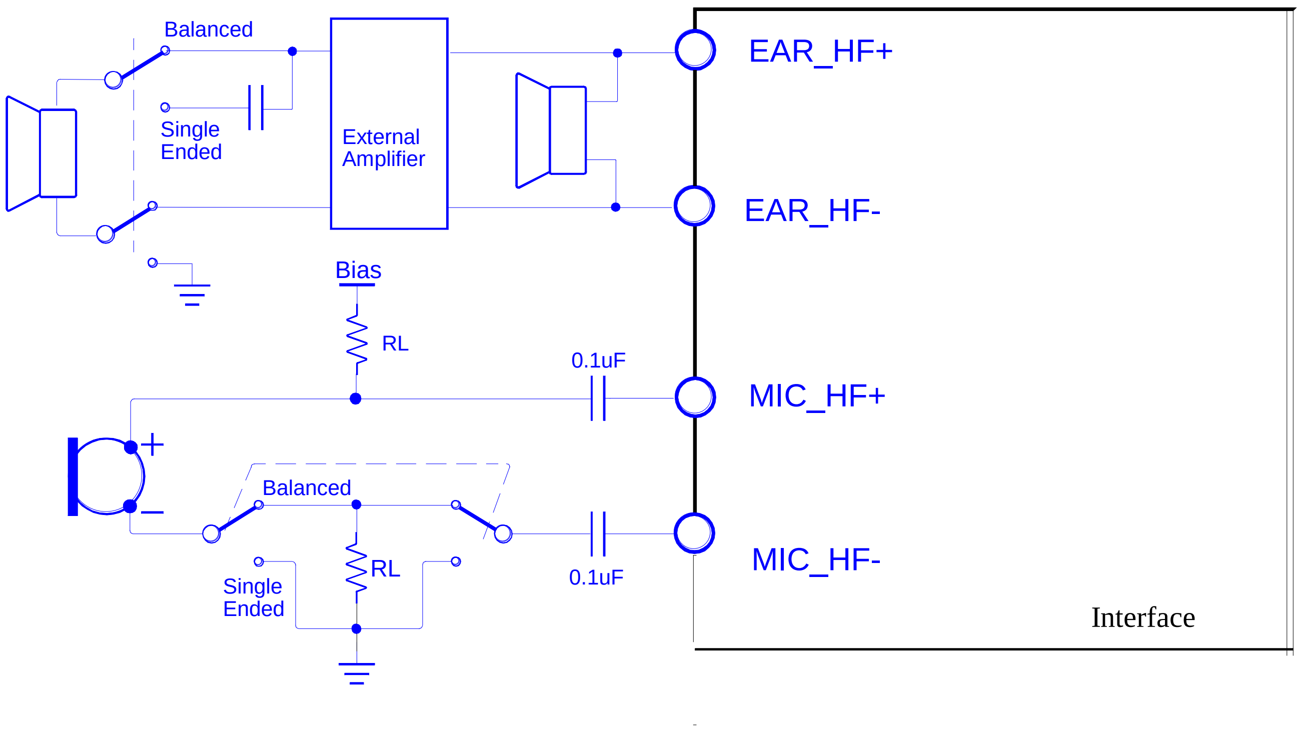

Figure 1. xE910 Available solution to connect external acoustic transducers to ***Audio Interface***.

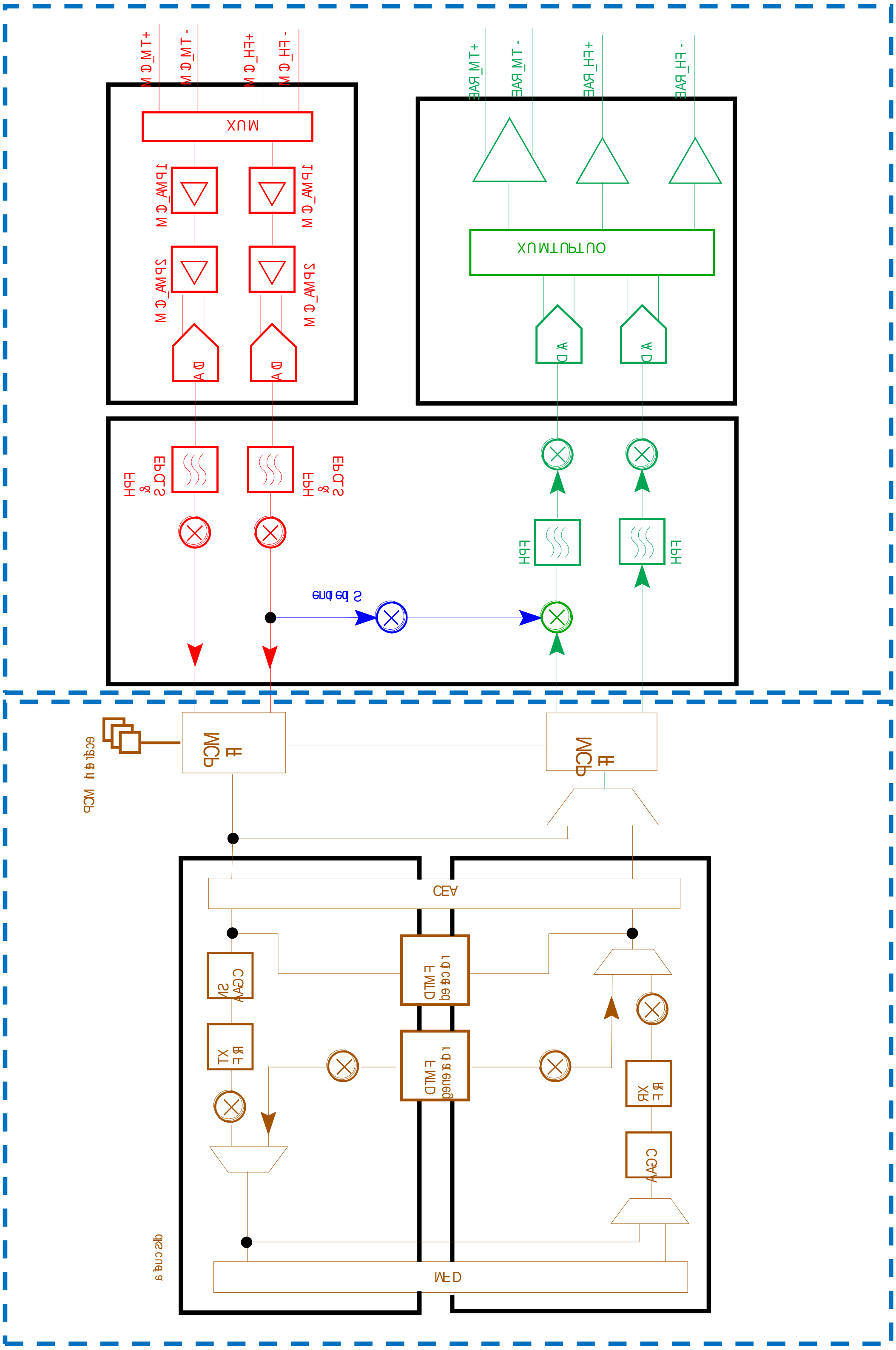

Figure 2. xE910 - Simplified block diagram of the analog part of the ***Audio Interface***

3. Why HS, MT and HF

Before explaining the acronyms HS-MT-HF, it is useful to make a short description of the acoustic transducers.

3.1. Terminology

Transducer is a device -usually electrical, electronic, or electromechanical- that converts one type of energy to another for various purposes, including measurement or information transfer.

Handset *(or* ***MicroTelephone****)* is the combined hand-held unit of microphone *(transmitting capsule)* and earpiece/headphone *(receiving capsule)* , particularly suitable for use in connection with a standard telephone. The spring-loaded hinge ensures that the headphone from outside fits tightly to the human ear and at the same time holds the microphone in the correct position.

← Figure 3. *Handset* is the transceiver for the audio on a *wired telephone*

Figure 4. *Handset* is the audio and radio transceiver on a *mobile phone* →

Transceiver is a device that has both transmitter and receiver, which are combined , sharing common circuitry or a single housing.

Headphones are a pair of transducers that convert an electrical signal into audible sound waves, generally used to prevent other people from hearing the sound either for privacy or to prevent disturbance *(e.g. in a public library)*.

Figure 5. Headphones.

Earphones *(or* ***Earbuds****)* are smaller sized *Headphones*, placed directly outside of the ear canal, but without fully enveloping it .
In the 1990s *Earbuds* became a common type bundled with personal music devices.

Figure 6. The headphones included with →
the *iPod*® are *Earbuds*

Headsets are also commonly understood to refer to a combination of Headphones with an attached Microphone, used for two-way communication *(for example with a mobile phone)*.

Earpiece is a small dimensions transducer that fits in *(hearing aid)* or is held next to the ear *(telephone receiver);* it is composed by a speaker and its holder.

Figure 7. The Earpieces had big dimensions at the beginning of telephony. At left : a model common around the turn of the 20th century.

Handsfree is the name of equipments that can be used *"without hands"* during a call. Not only, actually have they become also wireless by Bluetooth technology. Handsfree could be portable or fixed.

Portable . Used when talking by mobile phones , in some countries its use is mandatory while driving.
Generally the user can answer the call by a single-touch button.

Figure 8. A stereo version of Portable Handsfree.

Fixed . In this case the telephone unit contains both a microphone and a loudspeaker separately from those in the Handset. It is used to transfer the input and output sounds from the Handset to the ambient, allowing several persons to participate in a call simultaneously without the telephone receiver being held.

A *Fixed Handsfree* kit fitted in the telephone unit is called **Speakerphone** .
It may be divided into *half-duplex* and *full-duplex* type:
the first one allows sound to travel in one direction at a time *(from the telephone line to its user or from its user to the telephone line)*, while the second one is able to transmit and receive simultaneously without discernible change of transmission direction.

← Figure 9. The *Office environment* version

A *Fixed* Handsfree kit fitted into a car is called ***CAR-KIT***.
It may be used by the driver and/or the passengers.
It is mandatory its use in the vehicles while driving.

Figure 10.The car environment version. →

3.2. Introduction

As said before, both in transmit and in receive sections the ***Audio Interface*** provides two audio paths, active only one at time. Every section comprises digital filter stages and buffering amplifiers, whose configuration could be *Differential* or *Single Ended* *(see paragraph 4)* .
Moreover the ***Sidetone*** functionality could be implemented by the amplifier fitted between the transmit path and the receive path.

To select the well-suited section, refer to paragraph *3.3 Selection mode*.
To know which are the suggested and requested electrical characteristics of the audio transducers, refer to paragraph *4.1*, *4.2* , *8.4.1* and *10.3* .

3.2.1. History

The *Baseband* chip has been developed for the cellular phones, which needed separated amplifier chains in *RX* and in *TX* section : one amplifier section had to be used with internal audio transducers while the other amplifier section must be used with external audio transducers.
To distinguish the schematic signals and the Software identifiers, two different definitions were introduced, with the following meaning:

- internal audio transducers → *HS/MT (from HandSet or MicroTelephone)*
- external audio transducers → *HF from (HandsFree)*

as exposed in the previous paragraph *3.1 Terminology* .

3.2.2. Nowadays transducers

For obvious reasons we have not changed the ***HS*** and ***HF*** acronyms, keeping them in the Software and on the schematics.

NOTICE . We want to remark that with the Telit modules <u>this distinction is not necessary</u> because the two sections:

- have fully equivalent electrical performances (like the two Microphone amplifiers)

- activate the same functionalities (like the Acoustic Echo Canceller module)

- offer slightly different performances (like the two Speaker Buffering stages)

3.3. Selection mode

The activation of the requested audio section is made by software by **AT#CAP** command, turning on:

- an output stage to drive a low impedance earpiece and an input stage to drive the handset microphone when in HS/HF mode;
- an output stage to drive a high impedance headset and an input stage to drive the headset microphone when in HF mode.

The *SIDETONE* functionality is on at request in both modes.

Remember that if you don't have any load driving constraint *(like a speaker with an impedance coil lower than 32Ω)*, being the performances between the two blocks almost the same, the choice could be done in order to overcome the PCB design difficulties, without considerations to the electrical performances, but respecting the Baseband chip specifications.

4. Electrical Characteristics

4.1. Input lines

	xE910
Line coupling	AC (*)
Line type	Balanced / Unbalanced
Coupling capacitor	0.1uF
Differential input impedance	16 ~ 24KΩ @ PGA Gain=0dBv
Differential input voltage	1Vrms @ PGA Gain=0dBv

Table 1. xE910 ***Mic*** *differential microphone paths*

(*) ***DANGER***

AC means that the signals from the microphone have to be connected to input lines of the module through capacitors which value has to be 1uF. Not respecting this constraint, the input stage will be damaged.

DANGER

Because particular OEM applications need a single input line connection, a Single Ended configuration can be implemented, thus halving the useful microphone signal level : it is forbidden the direct connection of the unused input to the Ground, but only through a 100nF capacitor.

In both cases the OEM circuitries shall be carefully designed to reduce the common mode noise typically generated on the ground plane.

4.2. Output lines

During the design process, remember that there are slightly different electrical performances when the load is driven directly from the internal audio amplifiers :

- The xE910 *Ear* lines can directly drive a *32*Ω in Differential configuration

There is no difference if the amplifiers drive an external amplifier.

	xE910
Differential Line coupling	DC
Differential internal Output load resistance	32 Ω
Differential Output resistance	32 Ω
Signal bandwidth	150 ÷4000 Hz *(-3 dB)*
Differential Output power	< 63mW

Table 2. ***Ear*** *Differential Out*

NOTE . We suggest to drive the load differentially from both receive drivers, thus the output swing is doubled and the need for the output coupling capacitor is eliminated. However if particular OEM application needs , also a Single Ended circuitry can be implemented, thus reducing by four the output power.

The OEM circuitry shall be designed to reduce the common mode noise typically generated on the ground plane and to get the maximum power output from the device (low resistance tracks).

(*) DANGER

Using Single Ended configuration, the unused output line must be left open. Not respecting this constraint, the output stage will be damaged.

5. Single Ended or Differential

Audio amplifiers could be designed in two configurations: *single-ended* and *differential*.
The differential output power amplifier configuration is also called bridge-tied load (BTL).

In the next paragraphs we will discuss about the use of these two input and output configurations.

Also calculation and circuit examples will be exhaustively performed in the following chapter.

5.1. Concepts

Any voltage can be characterized by a potential difference between two terminals.

The configuration of the two terminals and how the signal is delivered from output to input allows the signal to be more generally described in one of three ways:

- *Single-ended signal*. This is a signal delivered between a signal track and a ground. One terminal for a single-ended connection is always at fixed potential (*usually Ground*).
- *Differential Signals*. These are signals that travel through a pair of tracks. On the signal pair, neither of the terminals is Ground.
- *Common mode Signals. They represent a special case of differential signals, also traveling between a pair of tracks, where the voltage potential on both signals is the same.*

5.1.1. Advantages

Differential amplifiers are desirable to use, especially in audio applications where the amplitude of the signals is very low, like output of the microphones. The benefits received from using differential amplifiers could be summarized as:

- *Increasing of Common Mode Rejection Ratio (CMRR).* Differential inputs enable cancellation of any noise common on both inputs. Noise generated at the input of the amplifier has a greater effect than noise generated at the output, because any noise on the input is multiplied by the gain of the amplifier.
- *Increasing of Signal to Noise Ratio (SNR).* The inputs to the amplifier are particularly sensitive to noise, because typically they are not driven by a very low impedance source.
- *High Rejection in Electromagnetic Interference (EMI).* Noise immunity is very important in wireless devices because the RF signal is sent in bursts, so that the frequency between two bursts falls in the audio bandwidth. The RF rectification is such a problem that many manufacturers have to strongly shield the audio part of their applications.
- *Doubling the Useful Signal Level* .The voltage swing to the load is doubled, and then the AF power to the load is 4 times the single-ended AF power at the same voltage supply.
- *No need of output blocking capacitor.* Even if the differential outputs are biased at half-supply, no DC voltage exists across the load: therefore you will lower the overall cost

and save PCB space, because you don't need the big, expensive and heavy blocking electrolytic capacitors (generally from 33µF to 1000 µF).

- *Wider bandwidth.* Due to the absence of blocking capacitors, the speaker impedance creates no frequency limiting effect.
- *Less shielding is required from amplifier to load*.

5.1.2. Disadvantages

The additional track routing could be very difficult and requires more board space but this is mainly the only one disadvantage while implementing a differential amplifier.

6. Microphone Amplifier

6.1. Single Ended (unbalanced)

The basic diagram of a single ended input microphone amplifier is shown below; it works as the name implies: the signal output is the ***Vin*** voltage multiplied by the amplifier gain ***A.***

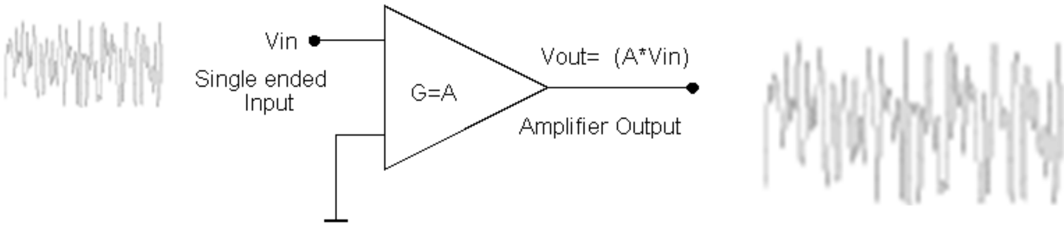

Figure 13. Single Ended input amplifier

With this configuration it is more difficult to obtain a good common mode rejection (CMRR) because any signal appearing on input terminal will appear multiplied at the output.
The advantage of having an unbalanced input device is that "*it has only one active signal*". Requiring less space on board, this configuration solves critical PCB dimensions problems but in most cases introduce the need of strong shielding.

6.2. Differential (balanced)

The basic diagram of a differential input microphone amplifier is shown below; it works as the name implies: the signal output is the difference between the ***In+*** and ***In-*** voltages, multiplied by the amplifier gain.

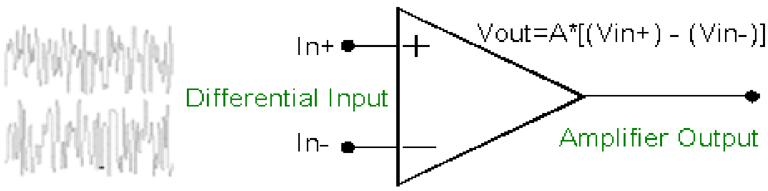

Figure 14. Differential input amplifier

It is easy to understand why a differential input amplifier provides common mode rejection (*CMRR*): only the difference between the input terminals is amplified, any signal appearing on both input terminals will not appear at the output.

The most important factor in achieving benefits from differential input circuitry is the PCB layout and not the choice of the transducer: differential tracks should always be run close together from microphone to amplifier and from amplifier to module.
Matching track impedances is very important for maximizing CMRR, increasing SNR and minimizing EMI.
All impedance elements should be added to both tracks of the differential pair and tracks should always be run over a ground plane to shield from EMI.
The fact that ECM uses a FET output stage that essentially acts as a current source makes it easier to connect its output differentially to produce a differential signal 6dB higher than single-ended connected device.

6.3. Suggestions

When using an amplifier with differential inputs, there are techniques to ensure that the device is configured correctly. If the source driving the amplifier is:

- *Single-ended*. Ground one amplifier input through a capacitor near the source of the other input. Grounding the capacitor near the source enables common-mode noise cancellation.
- *Differential*. It is very important to keep the same length for the differential tracks, close together to cancel any common mode noise induced in the trace itself.
- *Impedance should be matched between tracks, over audio frequencies as well as RF frequencies*

6.3.1. EMI protection

Since electromagnetic waves induce current when propagating in the presence of a conductor, *EMI* is almost always common mode in nature since the wavelengths are usually quite large in comparison to the spacing between the differential traces. The EMI-induced current translates to noise voltage and, if the track impedances are matched, this will yield common mode voltages that will be rejected by the differential amplifier.
It becomes quite clear that the following conditions should be observed to effectively reject EMI, when designing a PCB for differential input amplifiers:

- Differential tracks from the microphone to the amplifier should be traced close to each other, to maximize the common mode rejection for EMI.
- It is a good rule to space the differential tracks by one track width.

7. Speaker amplifier

The audio power amplifier is a critical component in your application; for this reason its configuration becomes very important.

7.1. Single Ended (S.E.)

The single ended signal output must be AC coupled to the load. Without the output coupling capacitor, the half-supply bias across the load would result in both increased internal IC power dissipation and also permanent loudspeaker damage.

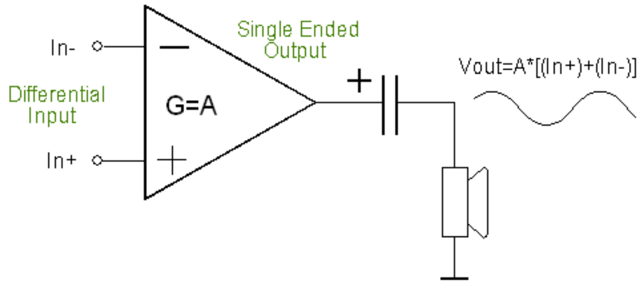

Figure 15. *Single Ended* output amplifier

Mainly we have only one advantage: having only one active track, the routing of signal line requires less board space.

7.2. Differential (BTL)

It provides a differential signal output, where the ***Vout-*** side is the mirror image of the ***Vout+*** side, without any AC coupling output capacitor.
In the differential drive, while one side of the amplifier is slewing up the other side is slewing down, and vice versa: the voltage swing on the load is doubled.

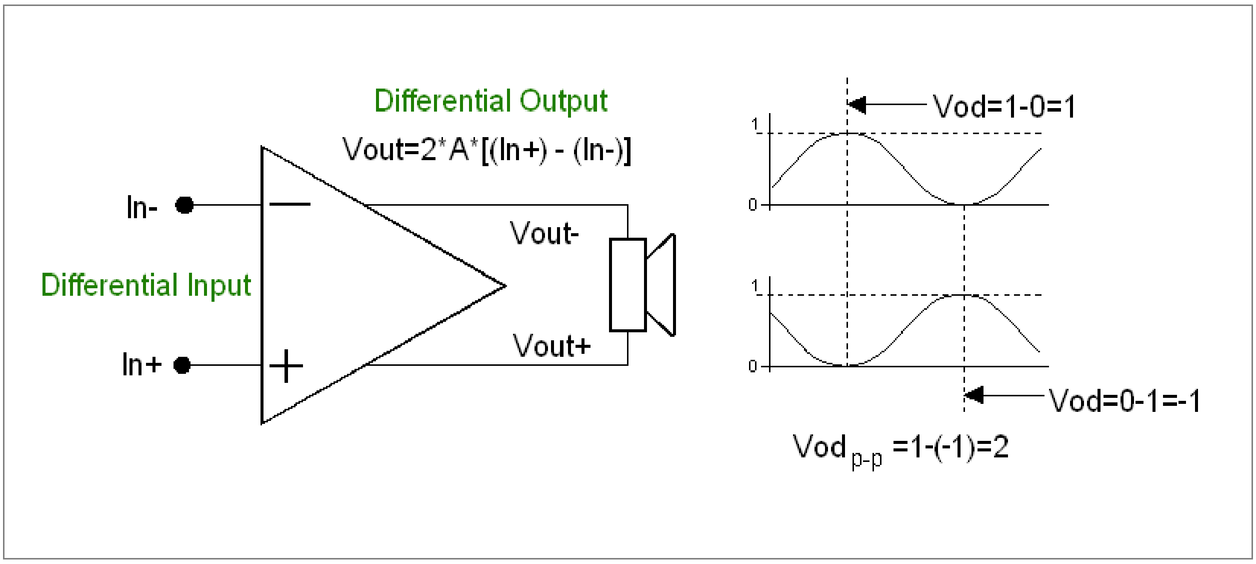

Figure 16. Differential output amplifier (BTL)

The differential drive configuration offers several potential benefits:

- *Doubling the voltage swing on the load*. The power to the load is 4 times the output power of a single-ended output at the same voltage supply.
- *No output blocking capacitor is needed*. Even if the differential outputs are biased at half-supply, no DC voltage exists across the load.
- *Cost and PCB space are minimized*. You don't need the big blocking capacitors (approximately from 33 µF to 1000 µF), which are expensive, heavy, and occupy wide PCB area.
- *Low-frequency performance* is limited only by the input network and speaker response. There is no frequency limiting effect due to the high pass filter network created by the speaker impedance and the coupling capacitance.
- *Less shielding* is required from amplifier to load.

7.3. Suggestions

The increase in output power assumes that the amplifier is not current limited or clipped. Choose the right amplifier closed-loop gain not to generate excessive clipping, which will damage high frequency transducers used in loudspeaker systems.

8. Electret Microphone

8.1. Generality

The **Microphone** is an acoustic to electric transducer (sensor) that converts sound pressure into an electrical signal. It is also called ***mike*** or ***mic***.
The ***Electret*** is a stable dielectric material with a permanently-embedded static electric charge, which will not decay for hundreds of years, due to the high resistance of the material. Its name comes from *electr*ostatic and magn*et* , drawing analogy to the formation of a magnet by alignment of magnetic domains in a piece of iron.
The ***Electret Microphone*** (shortly *ECM*) is a type of condenser microphone whose audio pickup section has a structure of a condenser consisting of a diaphragm and an opposite back plate. Usually a very high voltage (tens or hundreds volts) should be applied externally to polarize such a condenser. However, because the electric charge can be maintained in a polymer film by the *electret effect*, thereby the polarizing direct-current high voltage is eliminated. So it could offer the desired long-term stability and ultra-flat frequency response.

8.2. Principle of operation

The sound waves impinging on the diaphragm cause the capacitance between it and the back plate to change in sympathy.
Because the capacitance is relatively small, the electrical impedance is high and unmanageable; to overcome this, the condenser microphone incorporates a JFET to transform the impedance to a lower level suitable for feeding an amplifier via screened lead.

8.3. Types

There are three ***ECM*** technologies, according to where the electret film is used:

- *Foil-type*. The diaphragm itself is made of an electret polymer film. This is the most common type, but offering also the lowest quality, since the electret material doesn't make a very good diaphragm.
- *Back-type*. An electret film is applied to the back plate of the microphone capsule and the diaphragm is made of a superior, uncharged material.
- *Front-type*. In this newer type, the back plate is eliminated from the design, and the condenser is formed by the diaphragm and the inside surface of the capsule. The electret film is adhered to the inside front cover and the metalized diaphragm is connected to the input of the FET.

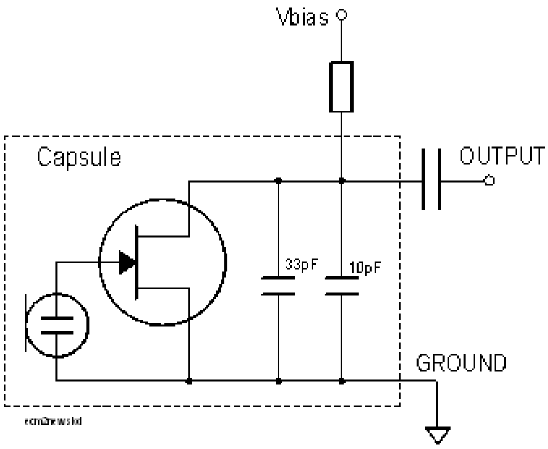

Figure 17. ECM schematic block

8.4. Performance

The ECM is particularly useful in the hand held and telecommunications field because it has the following advantages compared to the dynamic type:

- flat frequency response extended both at low and high frequencies
- immunity to vibration, due to the very low mass of the diaphragm
- low supply voltage
- low noise
- long-term stability
- smaller dimensions

8.4.1. Electrical Characteristics

The following table lists some typical ECM electrical characteristics.

Nominal sensitivity	-45dBV$_{rms}$/1Pa (+/- 3dB)
Line coupling	AC
Nominal Voltage	2V
Range of Using Voltage	(1÷10) V
Consumption Current	(150÷500) µA
Impedance	2,2KΩ
Signal to Noise Ratio	56dB @1KHz/1Pa (A curve)
EMI capacitor between terminals	10pF, 33pF

Table 5. Microphone electrical characteristics

8.4.2. Differential Connection Advantages

- *Increase in CMRR*. The differentially connected ECM does not improve CMRR alone. CMRR is defined as the ratio of the common mode input / output. Thus, common mode rejection is not a function of the ability to generate a differential output, but instead a measure of how well an amplifier rejects a common mode input. The tracks between the microphone and the codec are more important to CMRR than the way the signal is.
- *Useful signal***.** The signal level is 6dB higher.
- *SNR increase***.** The differentially connected ECM does not directly improve SNR either. While it might be argued that since the signal is increasing by 6dB, this improvement will also increase SNR, but this is not true. Since all condenser microphones are inherently single-ended devices, self-noise is generated in series with the signal output and both are referenced to ground. Therefore, converting the single-ended signal to a differential output signal effectively amplifies the Signal and the Noise equally, yielding no net gain in SNR.
- *EMI rejection***.** As with CMRR, converting a single-ended signal to a differential one cannot improve EMI rejection because EMI is induced upon the tracks on the PCB. By maintaining good PCB design techniques as discussed, EMI rejection is improved by matching impedances to insure that EMI is applied by common mode to the differential amplifier where it will be rejected.

8.4.3. Internal Amplifier

Till now the maximum sensitivity available from ECMs has been $-38dBV_{rms}/1Pa$. If higher level is requested, on the market you can find devices that have a low noise amplifier in the same holder, whose gain is generally adjustable from *0dB* to *+20dB*. But one more track has to be added to your PCB because also the amplifier power supply is needed.

8.5. Biasing Voltage

An ECM usually needs a biasing voltage to work properly: verify on microphone manufacturer Data Sheet its characteristics that could be different from that listed on table 5.

If you do not respect the load, voltage and current requirements, the performances vary.
The figure 18 shows how External Load Resistor and Power supply affect the sensitivity.

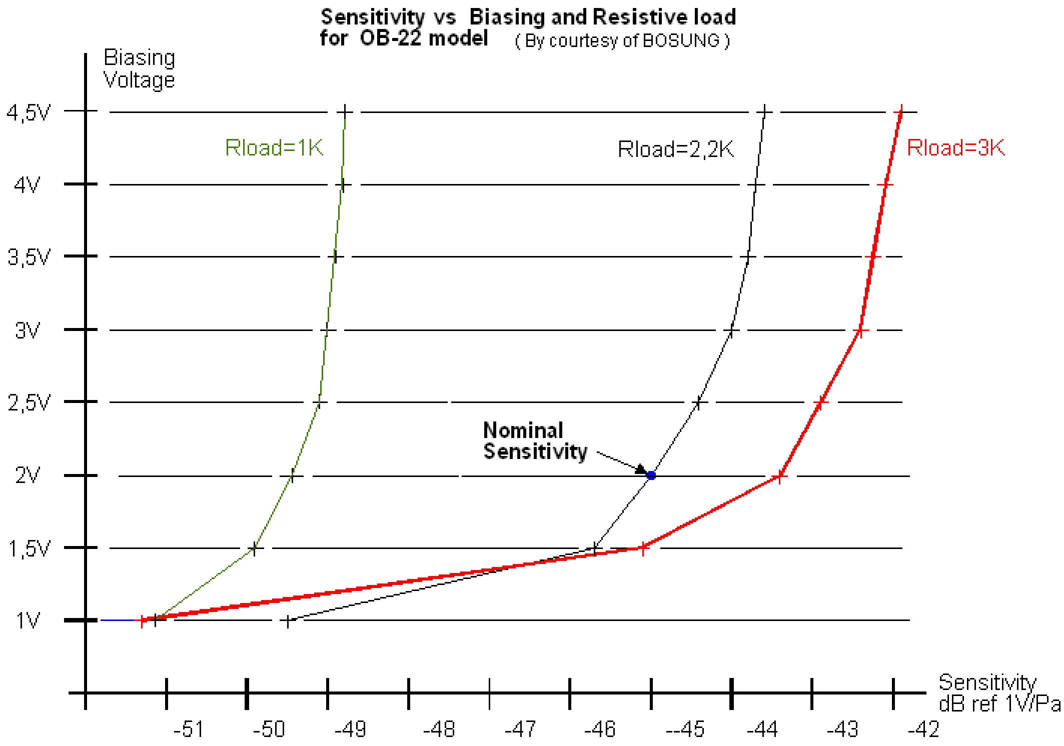

Figure 18. Sensitivity behavior of ECM model OB-22 (*By courtesy of BOSUNG*)

8.5.1. Biasing Source

The microphone could be powered from the ***Audio Interface*** itself or from a low noise external source.
Being the microphone circuitry the more noise sensitive, its design and layout must be done with particular care. Both microphone paths have been designed to implement the balanced configuration; if the OEM circuitry is balanced too, then the common mode noise typically generated on the ground plane is reduced.
However some customers strongly request the minus terminal of the microphone grounded and also an unbalanced circuitry can be used for OEM application.
Likewise the amplifiers two different power supply concepts are involved for the biasing:

- unbalanced biasing
- balanced biasing

8.5.2. Reminder

The ECMs have a hot wire were the positive biasing must be connected. Usually it is indicated by a plus (+) symbol or a red point ( ). If the polarity of the bias is reversed, then the microphone will not work properly.

WARNING:

Be sure to respect the microphone biasing polarity.

8.5.3. Unbalanced Biasing

The microphone powering refers to a common ground plane or the analog ground, hence the supply might suffer on ground noise injection into the microphone signal.
Despite this method is always possible, it is not really recommended because it requires the following:

- a real star connection in the ground plane, not always easy to realize
- a very strong low pass filter in the microphone supply, that means a small series resistor (to have low dropout) and a very big capacitor.

8.5.4. Practical Suggestions

With reference to figure 19, that is an example of the unbalanced microphone biasing circuit, apply the following suggestions if it is possible:

- get the supply biasing voltage from a dedicated voltage regulator (*U1*), in order to eliminate the noise present on the power lines; this regulator can be the same for all the audio sections.
- drive the microphone by a capacitor multiply circuit (*R1*-*Q1*-*C2*)
- use a *shielded* cable if the microphone is wire connected
- capacitor *C3* shall be *≥200nF* otherwise the frequency response will be cut at frequencies lower than 300Hz)
- capacitor *C3* must be placed close to the ***MIC_HF-*** or ***MIC_MT-*** pad or when possible close to the shielded cable connector.

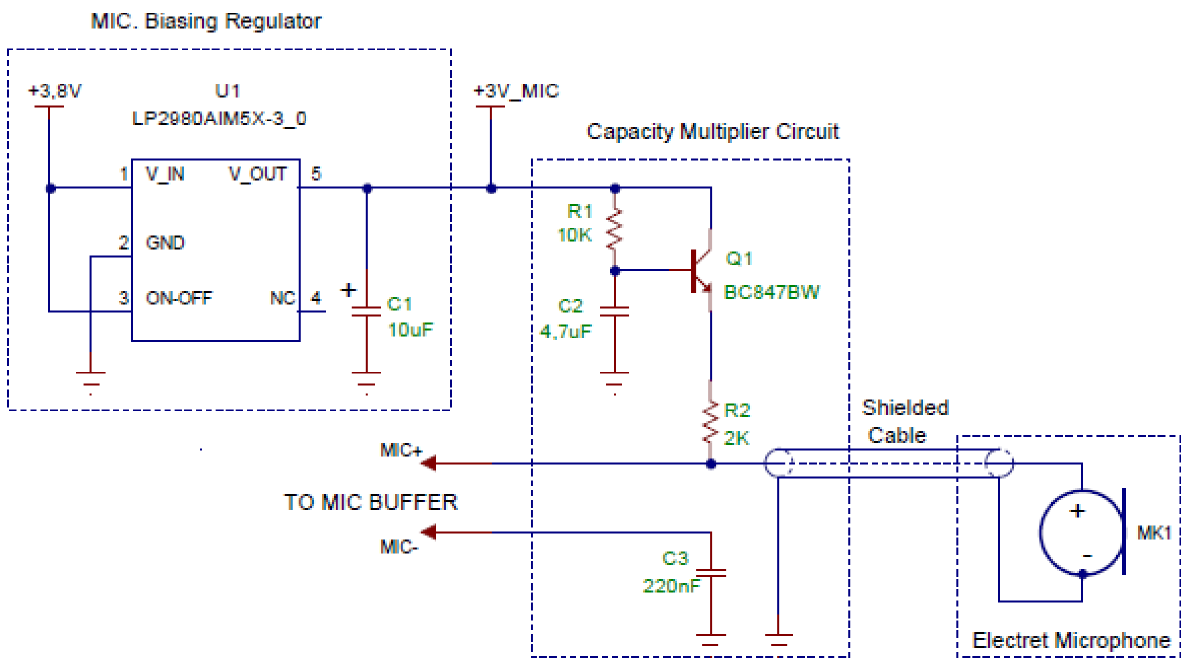

Figure 19. Example of unbalanced microphone biasing

8.5.5. Balanced Biasing

In this configuration, the microphone is not directly grounded but it is rather floating, with two symmetrically split load resistors.
This method gives benefits for external noise coupling due to symmetry: the noise coupling is only common mode signal and it is cancelled at the inputs of the low noise amplifier.
We highly suggest the *balanced configuration* even if it means one more track; in fact you need two active tracks, instead of one plus ground as in the unbalanced circuitry.

8.5.5.1. Suggestions

With reference to the figure below, that is an example of the balanced biasing circuit, apply the following suggestions :

- get the supply biasing voltage from a dedicated voltage regulator (*U1*), in order to eliminate the noise present on the power lines; this regulator can be the same for all the audio sections.
- drive the microphone by a capacitor multiply circuit (*R1-Q1-C2*)
- use a *twisted* cable if the microphone is wire connected
- split the nominal load in two halved value components
- try to keep the sum of R2 and R3 near the *Rload* declared in the manufacturer Data Sheet

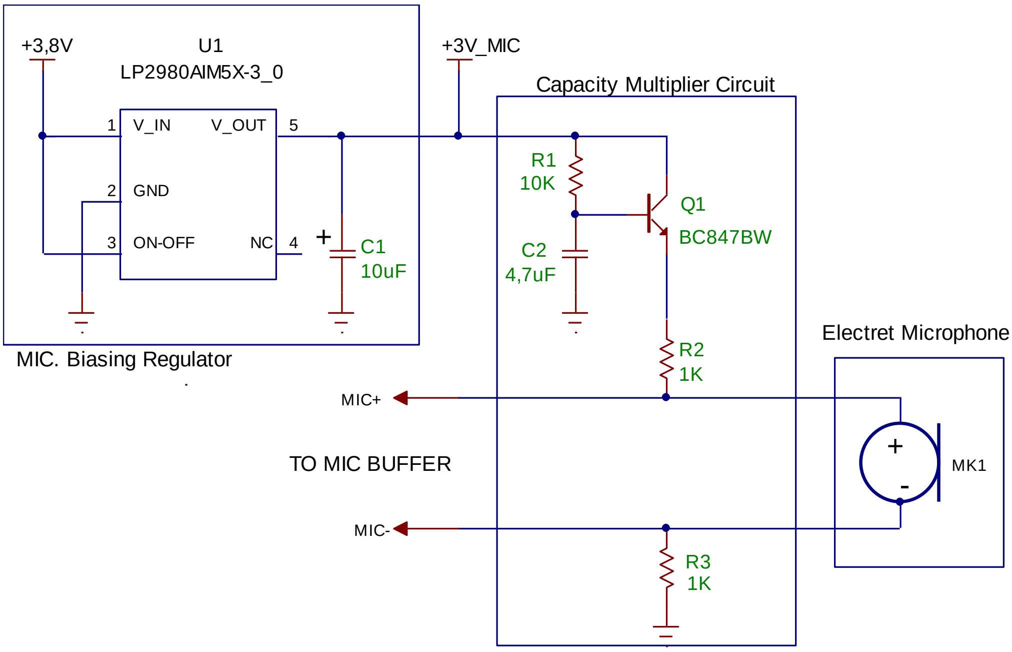

Figure 20. Example of differential microphone biasing supply

9. Input Paths Guide Lines

9.1. Reminder

As discussed at paragraph 4, there are several configurations for the input audio paths, but the best solution is the differential one, that uses 2 tracks to connect the microphone to its amplifier.
It is highly recommended to keep the whole microphone paths balanced.

9.2. Noise immunity

It is very important to have high noise immunity, because the frequency between bursts is 216Hz and falls in audio bandwidth. Because of the RF rectification, the generated disturbance becomes sometimes quite impossible to take off without strong shielding of the audio part of the PCB.
The best microphones contain two small value capacitors inside (*one for 900MHz and the second for 1800-1900MHz band*), which act as RF bypass to short-circuit the RF frequency components to ground, avoiding rectification phenomena.
Not only, the “source” pin is directly connected to its shield, and a lot of ground vias form a sort of shielded case.
When the screening action is not enough, you will hear a sort of noise we call Buzzing, because it seems to be a bug with its buzz sound, whose level depends from TX power.

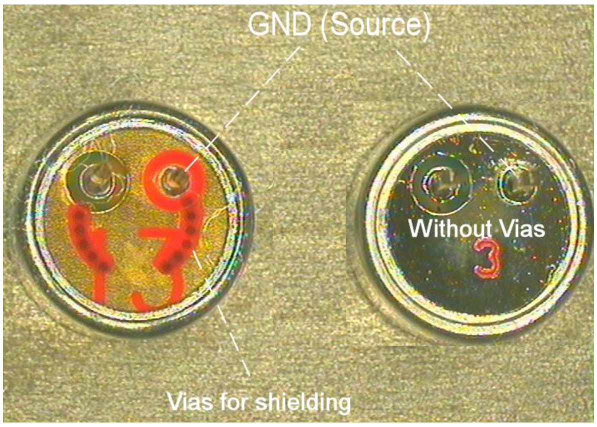

Figure 20 bis. At left the best shielded electret microphone.

9.3. Practical suggestions

- The AC coupling capacitors in the ***In+*** track should also be used in the ***In-*** track.
- Any shunt capacitors or series ferrite beads used for RF rejection should be applied equally to both ***In+*** and ***In-*** tracks.
- Any bias resistors or load resistors needed for differential source should be mirrored for both differential tracks.
- Ground tracks or a ground plane should be placed close (ideally beneath) the differential tracks. This provides a direct ground path for RF signals.
- The biasing circuit and eventually the buffer can be designed in the same way both for the internal and external microphones
- If possible use balanced (*differential*) microphone connection
- Keep the microphone tracks on the PCB and wires as short as possible.
- If your application requires an unbalanced (*single ended*) microphone connection, then keep the lines on the PCB balanced and "unbalance" the path close to the microphone connector if possible.
- For the microphone biasing voltage use a dedicated voltage regulator and a capacitor multiply circuit.
- Make sure that the microphone tracks in the PCB don't cross or run parallel to noisy tracks (especially the power line)
- use a rubber grommet to avoid the direct contact between microphone and its holder
- as a good rule , the hole in front of microphone should have 1mm diameter
- If possible, put a ground trace connected to the ground plane by several vias all around the microphone lines. It is advisable the use of “blind holes” as shown in the next figure. This physical arrangement simulates the shielded of the traces on the PCB.

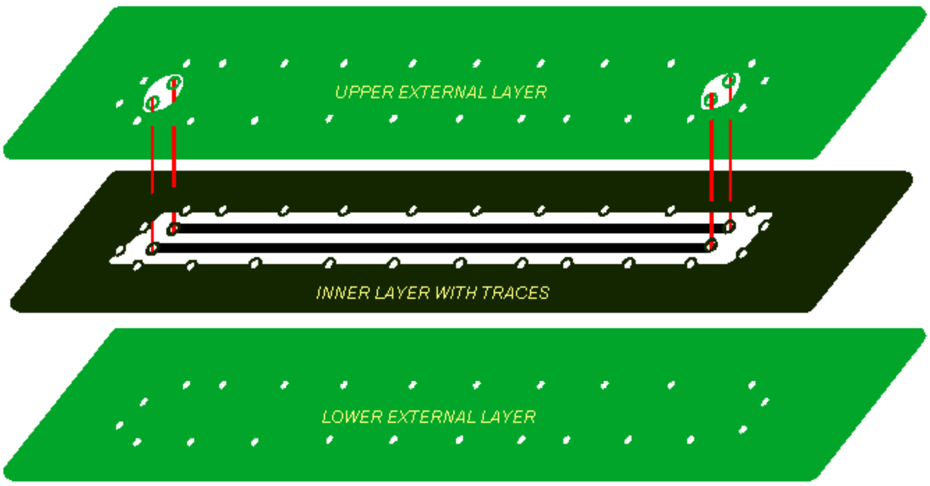

Figure 21. Example of “*coaxial like*” microphone realized by shielded tracks

9.3.1. Hands Free

When your application is working in an Hands Free system and the microphone is fitted in the same box with the speaker:

- try to have the maximum possible distance between them; if it is possible at least 7cm
- if you use an omni-directional type (and this is the typical application) please seal it on the rear side (no back cavity) in order not to collect unwanted signals ; try to make divergent the main axes of the microphone and speaker
- if you use an external microphone amplifier, set the module to the minimum possible gain

9.4. Definitions

9.4.1. Normal Spoken Condition

For a cellular phone, the ***normal spoken conditions*** take place when the talker mouth is about *7cm* far from the microphone. Under these conditions, the voice will produce an acoustic pressure of ***-4,7dBPa*** *@1kHz* on the microphone membrane.
Obviously, during a call this level varies, according to the volume of the talker voice. Usually you may apply the following rough rule of thumb to define the useful dynamic range:

- the *strongest voice level* condition is when the talker is screaming: the signal increases by +20dB
- the *lowest voice level* condition is when the talker is whispering: the voice level decreases by –50dB.

These limits must be considered for designing the external microphone amplifier.

9.4.2. Sensitivity and electrical equivalent level

The *Nominal Sensitivity* is expressed in *dBV/Pa* or in *dBV/μbar* and could be defined as:

"*the output voltage level for a specified acoustic stimulus under specific load condition*"

Because *0.1Pa = 1μbar*, the difference between one measuring unit and the other is:

Δ=20 $\log_{10}$ 0.1= **- 20dB** as example: -40*dV/Pa* = -60*dBV/μbar*

Generally, the nominal sensitivity is reported on Manufacturer Data Sheet.

9.4.3. Microphone Electrical Level

Knowing the nominal sensitivity of a microphone, it is possible to calculate the voltage level on its pins under "*normal spoken*" condition.
A microphone having the nominal sensitivity of *-45dBV$_{rms}$/Pa* will produce an electrical equivalent signal at its pins:

Mic Level $= (-45) + (-4.7) = \mathbf{-49.7\ dBV_{rms}}$ or ***Mic Voltage*** $= 10^{(-49.7/20)} = \mathbf{3.3 * 10^{-3}\ V_{rms}}$

9.5. Microphone connections

The microphone could be connected whether directly or through a buffer amplifier to the input lines of the module. In the first case the required gain will be set only in the internal ***Audio Interface*** amplifier stages; in the second case, the required gain will be split between the external amplifier (G_E) and the internal stages (G_I) .
Again the external buffer amplifier can be ***fully differential*** or ***single ended*** configured: where possible it is always preferable a *fully differential* configuration (*balanced*). The buffering circuit shall be placed close to the microphone or close to the microphone wire connector.

9.5.1. Coupling

When using microphones with codecs that utilize differential inputs, special attention needs to be paid to the DC bias of the input signals. Since op-amps in differential configuration amplify DC as well as AC signals, bias differences are important. In the case of DC, common mode rejection means DC removal when the DC bias of the differential input signals is the same.
However, a mismatch in bias levels can quickly lead to reduce dynamic range since the output bias will be affected by the input bias difference.
Luckily, many codecs manufacturers provide amplifier biasing internally.

When internal biasing is used, DC matching is trivial. AC coupling both differential inputs tracks removes DC bias from both tracks, leaving the codec to provide the appropriate bias for best dynamic range. The figure below shows AC coupled differential inputs connection.

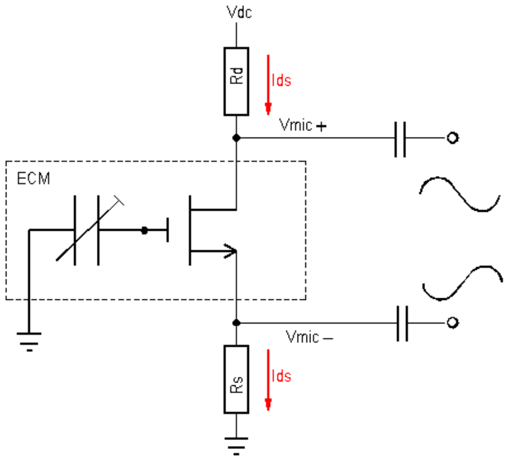

Figure 22. ***Fully differential*** ECM ac-coupled output signals configuration

9.5.1.1. Practical Suggestions

As previously explained, when the user application is a *handsfree* system, the microphone and the speaker are far from module, driven by external amplifiers. In these environments when designing the external microphone amplifier you must take into account the voice attenuation, due to the distance between the talker and the microphone itself.
You must consider that ambient noise will be picked up.
To overcome these problems, it is preferable to set the gain of the microphone *10dB* lower with respect to the calculated value for a nominal sensitivity. An increased voice volume of the talker, which will speak louder because of the ambient noise, will compensate the corresponding reduction in signal level.
Usually the distance between the microphone and the talker in a *car cabin* is *40/50cm*; in these conditions the attenuation can be considered as a thumb rule around *20dB*.
The same considerations can be made for the *HeadSets* having the microphone on the earpiece cable (like a mobile phone).
The *HeadSets* having the microphone sustained close to the mouth can be threatened as a HandSet.

9.5.1.2. Losses Compensation

The voice signal, that in the "*normal spoken*" conditions produces on the microphone membrane an acoustic pressure of *-4,7dBPa* at 1kHz, will have a further attenuation of *20dB* due the *50cm* distance. A microphone having a sensitivity of -45dBV$_{rms}$/Pa will produce a signal:

$$MicLevel = (-45) + (-4.7)-20 = -69.7\ dBV_{rms}$$

equivalent to : $MicVoltage = 10^{(-69.7/20)} = 0{,}33 * 10^{-3}\ V_{rms}$

If the external amplifier gain G_E*=+10dB (3 times),* at the *normal spoken* conditions the signal will be:

$$Mic\_Input\ Level = 0{,}33 * 10^{-3} * 3 = 1 * 10^{-3}\ V_{rms}$$

at the ***Audio Interface*** input lines , resulting 3 times lower than the nominal as suggested .

These considerations are valid for both ***Mic_MT*** and ***Mic_HF*** input lines, if the same configuration is implemented.

9.5.2. Maximum Gain

When you define the gain of the microphone path, keep in mind that the maximum differential level to ***Audio Interface*** input lines must be ≤ ***1,03V***$_{pp}$***/-8,67dBV*** (*see paragraph 3.1*),if the internal gain G_E is set to 0dB.
We have to consider two different configurations: with or without external amplifier.

9.5.2.1. With External Amplifier

Let's calculate the external amplifier gain G_E at strongest voice level condition and internal amplifier gain G_I **=0dB**:

$$\left[(MicLevel + 20dB) + G_E + G_I\right] \leq -8{,}76dBV \longrightarrow \left[-49{,}7 + 20 + G_E + 0\right] = -8{,}76$$

then $G_E = (49{,}7 - 20 - 8{,}76)dB = +20{,}94dB \longrightarrow$ G_E**= +20dB** available by commercial values

9.5.2.2. Without External Amplifier

Let's calculate the internal amplifier gain G_I at strongest voice level condition and***:***

$$\left[(MicLevel + 20dB) + G_I\right] = -8{,}76dBV \longrightarrow \left[-49{,}7 + 20 + G_I\right] = -8{,}76$$

then $G_I = (49{,}7 - 20 - 8{,}76)dB = +20{,}94dB \longrightarrow$ G_I **= +21dB** set by dedicated AT command

9.5.3. Single Ended (Unbalanced) External Amplifier

The next figure shows a possible single ended input microphone amplifier

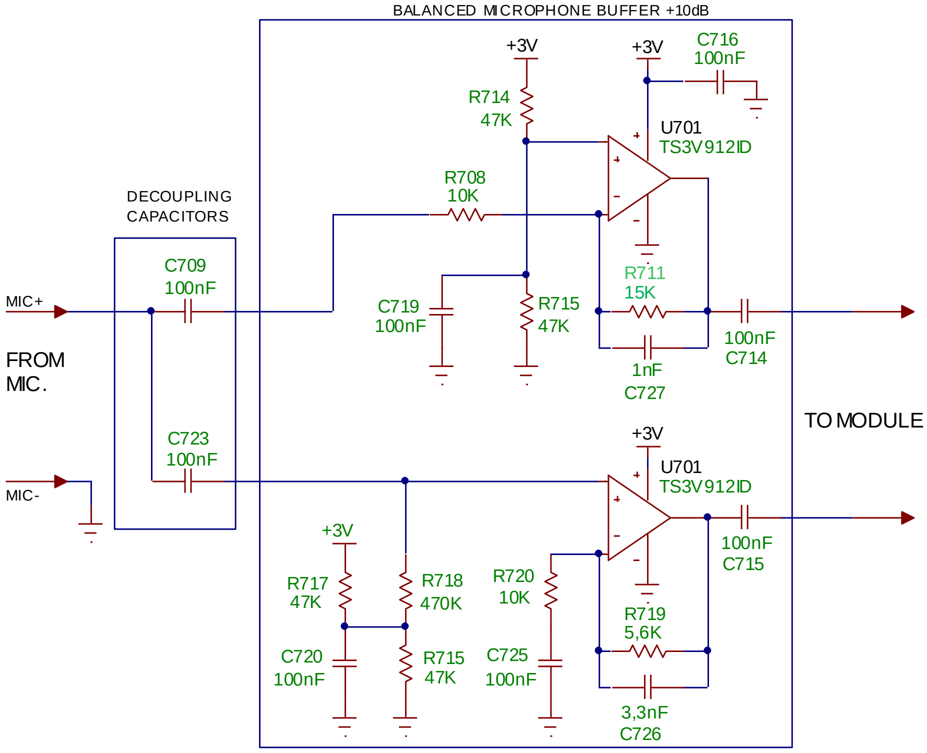

Figure 23. Example of Single ended input microphone amplifier

9.5.3.1. Practical Suggestions

- The gains of the two amplifiers are given by the formulas:

$$Gain(\text{NOT INVERTING BUFFER}) = 1 + \frac{R719}{R720} \qquad Gain(\text{INVERTING BUFFER}) = \frac{R711}{R708}$$

Assigning half of overall gain to each amplifier, you will obtain the requested one because of doubling the microphone signal path; in fact by the use of two amplifiers (the upper as "*inverting*" and the lower as "*not inverting*" configuration) we obtain an additional *+6dB* gain (*2 times*).

Remember: the "not inverting" amplifier section gain shall not be less than 1.

The amplifier overall gain can be modified changing the value of resistor R719/R720/R711, and the capacitors C726/C727as a consequence.
It is advisable to change R708 **only** if you have difficulty to find a commercial value for R711; in this case change R708 as little as possible because it acts as the input resistance.

- The buffer bandwidth at -3dB shall be 4KHz.
- Considering C725 >> C726, the -3dB bandwidth is given by the approximated formula:

$$freq. = \frac{1}{2\pi * R719 * C726} = \frac{1}{2\pi * R711 * C727} \quad [\text{Hz}]$$

first set C276 and C277 at commercial values; then choose R719 and R711 nearest to the commercial values

- The biasing for the *inverting* section is given:
  by the series divider *R714-R715* plus the capacitor *C719* , needed to filter the noise that could be coupled to that divider.

- The biasing for the "*not inverting*" section is given:
  by the series divider *R715-R717* plus the capacitor *C720* , needed to filter the noise that could be coupled to that divider , through a series resistor *R718* .

- This schematic does not include the required biasing circuitry for the microphone
- Decouping capacitor on ***MIC-*** line is not needed (see figure 19, capacitor C3)

9.5.3.2. Calculus example

Let's assume that:

- you are developing a *HandsFree* equipment
- your microphone have a sensitivity *of* $-45dBV_{rms}/Pa$ in "normal spoken" conditions at acoustic pressure of *-4.7dBPa*
- you want to use the 2nd differential microphone path *("**Mic_HF**"* input lines)
- the buffer amplifier have a gain $\boldsymbol{G_E}$=+20dB (*10 times*)
- the desired cutoff frequency of the buffer amplifier is $\boldsymbol{f_c}$ =4KHz
- the microphone is about 50cm far from the mouth of the talker

Due to the distance between the mouth and microphone the signal loss will be 20dB loss.
As calculated in paragraph 9.5.1.2 the electrical level output will be:

$$MicLevel = (-49.7) + (-20) = -69.7\ dBV_{rms} \quad or \quad MicVoltage = 10^{(-69.7/20)} = 0{,}33*10^{-3}\ V_{rms}$$

In the same paragraph it is suggested to increase the level to $1mV_{rms}$ by the external buffer, which will be designed with a gain:

$$G_E = \frac{Vinput\_lines}{MicVoltage} = \frac{1*10^{-3}}{0{,}33*10^{-3}} = 3 \longrightarrow \text{or } G_E = +10\ dB$$

To calculate the resistor values it must be kept in mind that "*balancing the line will double the signal*" and hence +6dB are already added.
Therefore the buffer stage must gain only 1,5 times or $\boldsymbol{G_E}$ **=+3,52dB**.
Keeping the input resistance =10KΩ, the corresponding values for the resistors on the buffer could be:

R711 = $\boldsymbol{G_E}$ * R708= 1.5*10 =**15kΩ**
R719 = ($\boldsymbol{G_E}$ -1) * R720 = (1.5 -1)*10 =**5kΩ**

The commercial values of *15kΩ* and *5.6kΩ* could be used.
As a consequence of the assigned resistor values, the nominal values of C726 and C727 should be:

C726= 1/ (2π*4000*R719)= 7.10 *10 $^{-9}$ F
C727= 1/ (2π*4000*R711)= 2,65 *10 $^{-9}$ F

rounded off at *6,8nF* (f_{cl}=*4181Hz*) and *2,7nF* (f_{cu}=*3931Hz*) commercial values.

9.5.4. Fully Differential (Balanced) Buffering

The next figure shows a possible fully differential input amplifier.

This circuit has a gain G_E *=15* (*almost +24 dB*) and could be applied both to "***Mic_MT*** " and "***Mic_HF"*** input lines. The gain adjustment shall be done by changing the resistors R604 and R606 and as a consequence the capacitors C636 and C637 to maintain the bandwidth 150-8000Hz (*at -3dB*).
If the required value for R604 and R606 is not a standard one, you can change R605 and R607 as little as possible because they act as the input resistances.

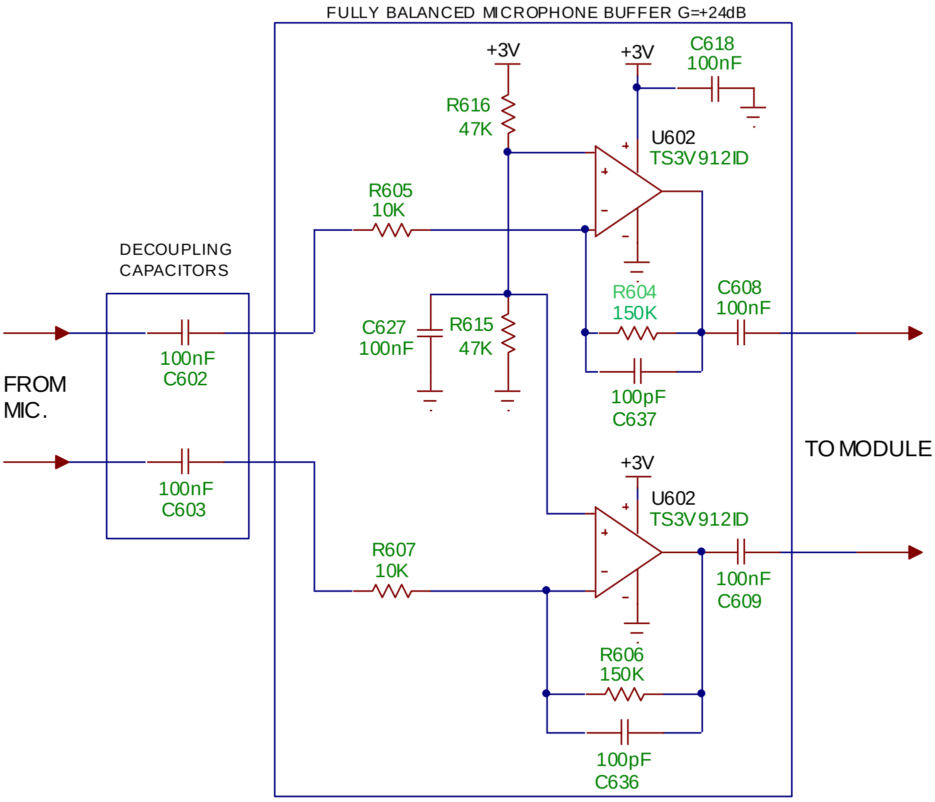

Figure 24. Example of the fully differential microphone amplifier

The buffer gain is given by the formula:

$$Gain = \frac{R604}{R605} = \frac{R606}{R607}$$

The C636 and C637 capacitors are placed in order to cut off the gain at higher frequencies than the transmitted GSM band; the cutoff frequency (*-3dB*) should be 8000Hz in order to have -1dB at 4KHz.
The cutoff frequency is given by the formula:

$$fcu = \frac{1}{2\pi * R604 * C637} = \frac{1}{2\pi * R606 * C636} \text{ [Hz]}$$

9.5.4.1. Calculus example

Let's assume that:

- you are developing a HandSet application
- you have a microphone with a sensitivity of *-50dBV$_{rms}$/Pa* in "normal spoken" conditions at acoustic pressure of *-4.7dBPa*
- you want to use the 1st differential microphone path ("***Mic_MT***" input lines)
- the buffer amplifier have a gain **G_E**=+24dB (15 times)
- the desired cutoff frequency of the buffer amplifier is **f_c** =4KHz

The output level from the microphone will be calculated as described in the paragraph 8.4.3:

MicLevel = (-50) + (-4.7) = -54.7 dBV$_{rms}$ or *MicVoltage = $10^{(-54.7/20)}$ = 1.84* 10^{-3} V$_{rms}$*

When the talker is screaming, the microphone signal will increase by 20dB (10 times) to *18,4mV$_{rms}$.*
Due to external amplifier, the level at the "*Mic_MT*" input lines will be:

$$Mic\_MT = MicVoltage * G_E = 18{,}4 * 10^{-3} * 15 = 276mV_{rms}$$

lower enough respect the maximum differential input voltage of *1,03V$_{pp}$/365mV$_{rms}$* as listed at paragraph 3.1 .
Choosing the input resistance *R605 = R607=10kΩ,* we will obtain the nominal values for the feedback resistors of the buffer:

*R605 = R607 = **G_E** * R607= **G_E** * R605 = 10*15kΩ = **150kΩ***

corresponding to available commercial values .

As a consequence the values of the capacitors *C636* and *C637* shall be:

$$C636=C637= 1/ (2\pi*4000*R606)= \mathbf{265 *10^{-12} F}$$

A commercial value of *270pF* gives a cutoff frequency of *3931Hz*, with an error less than 1,8%.

9.5.5. AFE Mic GAIN PARAMETERS

The differential input signal from the microphone passes a low noise amplifier with gain settings ranging from 0 to +42dB. Gains of the input lines can be adjusted separately, so that a maximal degree of flexibility is achieved, by dedicated AT commands described at paragraph 5.6.12

9.5.6. Transmit Block Diagrams

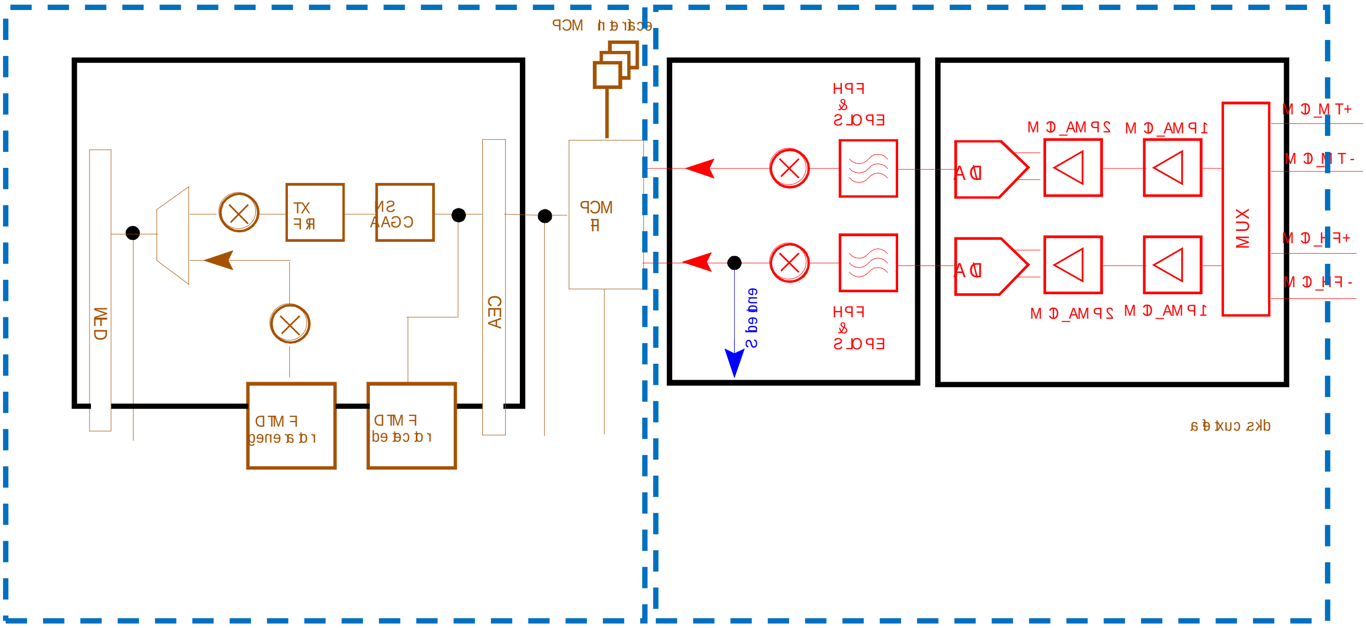

Figure 25 . xE910 Families TX section diagram

10. Output paths guidelines

10.1. Reminder

As suggested at paragraph 4, the differential configuration (*BTL*) is the best implementation for the internal output buffer amplifiers, even if it needs 2 tracks to the load.
Obviously, you must respect the electrical characteristics listed at paragraph 4.2.

10.2. Practical suggestions

All the designs shall comply with the following guidelines:

- Where possible use differential configuration (*BTL*) circuitry, to achieve the maximum power output from the device
- Keep the output tracks on the PCB and wires to the transducers as short as possible
- Make sure that the output tracks in the PCB don't cross or run parallel to noisy tracks (especially the power line)
- The cable to the speaker shall be a twisted pair with both the lines floating for the differential output configuration, shielded to ground for the single ended output one.
- If you want to implement a single ended output configuration, that directly drives the load without any external amplifier, leave one of the two output lines open and use only the other referred to ground. Remember that in this case:

a) *the output power is 4 times lower than the differential circuit and may not be enough to ensure a good voice volume.*

b) *you must use a big decoupling capacitor to the load , and this means more cost and wider PCB .*

- The I/O of the PCB should have a noise filter close to the connector, to filter the high frequency GSM noise. The filter can be a π type formed by an inductor of 39μH and 2 capacitors, one of *39pF (0603 case)* and the other of *1nF (0603 case)*

10.3. Mini Speaker characteristics

Generally both *HandSet* and *Portable HandsFree* use mini-speaker. In the below table you can find the main electrical suggested characteristics for such a transducers.

Rated Input Power	5mW
Maximum Input Power	20mW
Coil Impedance	32Ω @ 1kHz
SPL	95±3 dB @ 1KHz/1mW sine wave
Resonance frequency (Fo)	< 350Hz
Useful Bandwidth	Fo ÷ 8000 Hz @ -3dB

Table 6. Mini-speaker electrical characteristics

10.4. AF Power Requirements

There are several transducers that could be connected to output lines, but the various designs can be referred to three main categories, with different power requirements:

- handset
- portable handsfree
- fixed handsfree (car kit/speakerphone)

10.4.1. HandSet and Portable HandsFree

These devices have one mini-speaker inside that needs only few *mW* of driving power.
They can be directly drained from the Telit modules, provided a suited speaker is used, offering the cheaper and simpler solution, which will be adopted in most of the customer designs.

10.4.2. Fixed HandsFree (car kit/speakerphone)

These devices use a speaker that generally has a low resistive load and needs up to 5-10W of driving power, available only by external power amplifiers.

10.5. Speaker Connections

Because the M2M modules offer output balanced lines, when the differential configuration is implemented these output lines should be used to drive directly a speaker or as inputs to an external power amplifier. In such a way a higher common mode rejection ratio is obtained, reducing the GSM current bursts noise on the speaker output.

At low power constraints, you have two possibilities to connect the speaker: directly to module internal buffer amplifiers or through an external amplifier.
The choice will be made considering cost and performance, which generally clash.
At high power constraints, you must use an external power amplifier to boost the module output.

10.5.1. Direct connection

10.5.1.1. HandSets and Portable HandsFree

The direct connection is often the more effective cost solution, reducing the number of components to the minimum. But with some limitations:

- the speaker characteristics has to be almost exactly the suggested ones, otherwise the power output may be reduced (*speaker impedance bigger than 8Ω*) or the output amplification stage may be damaged *(speaker impedance lower than 8Ω*) .
- the reduced output power capability may not be enough for some particular applications.

10.5.1.2. Fixed HandsFree

These equipments require greater output:

- speakerphone needs at least 1W
- car kit needs at least 5W

Therefore the direct connection is not allowed.

10.5.2. External Amplifier

10.5.2.1. HandSets and Portable HandsFree

In this case the “***EAR_MT***” or“***EAR_HF***” lines from the *modules* should be AC coupled with a ceramic capacitor of 1uF.

The figure below shows the principle schematic of a differential configuration.

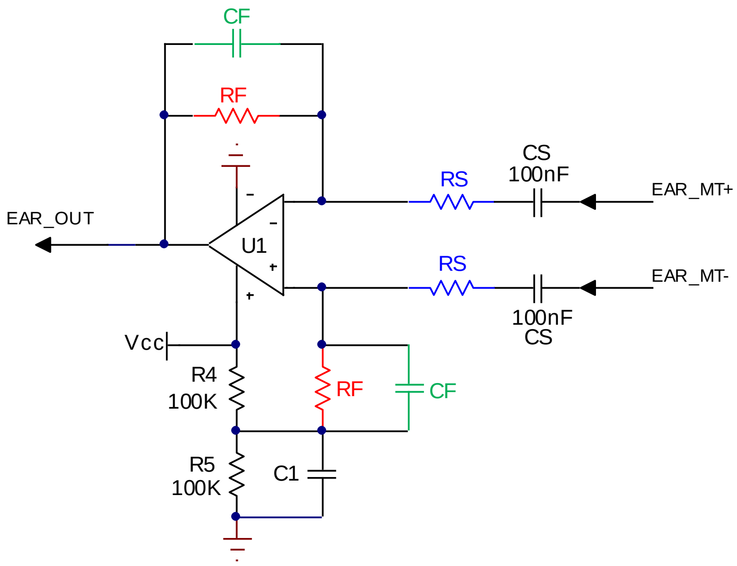

Figure 26. *Differential Output Amplifier* principle schematic

The resulting gain is: $Gain = \frac{RF}{RS}$

With corner frequencies : $fcl = \frac{1}{2\pi * RS * CS}$ [Hz] $fcu = \frac{1}{2\pi * RF * CF}$ [Hz]

Reminding that:

$$X_{C1} = \frac{1}{2\pi * fcl * C1} \text{ and } X_{C1} << RF \longrightarrow C1 \geq \frac{1}{2\pi * X_{C1} * fcl * 0.1 * RF}$$

The figures 27 and 28 show the possible schematic of two Audio Power Amplifiers external to ***Audio Interface,*** that have a mute control (*SHUTDOWN* pin) in order to turn it off while the device is not sending signal to the output; in this way the amplifier background noise is cut off ,avoiding to be audible during idle conditions .

Some amplifiers require a low impedance load at high frequency in order to avoid auto oscillation; this can be made with a capacitor in series with a resistor inserted between output lines and ground (box *Optional* in the figure 27)

When designing your application, remind to provide an adequate bypass capacitor to the amplifier, placing it close to the power input pin of the IC, to have the shortest traces.

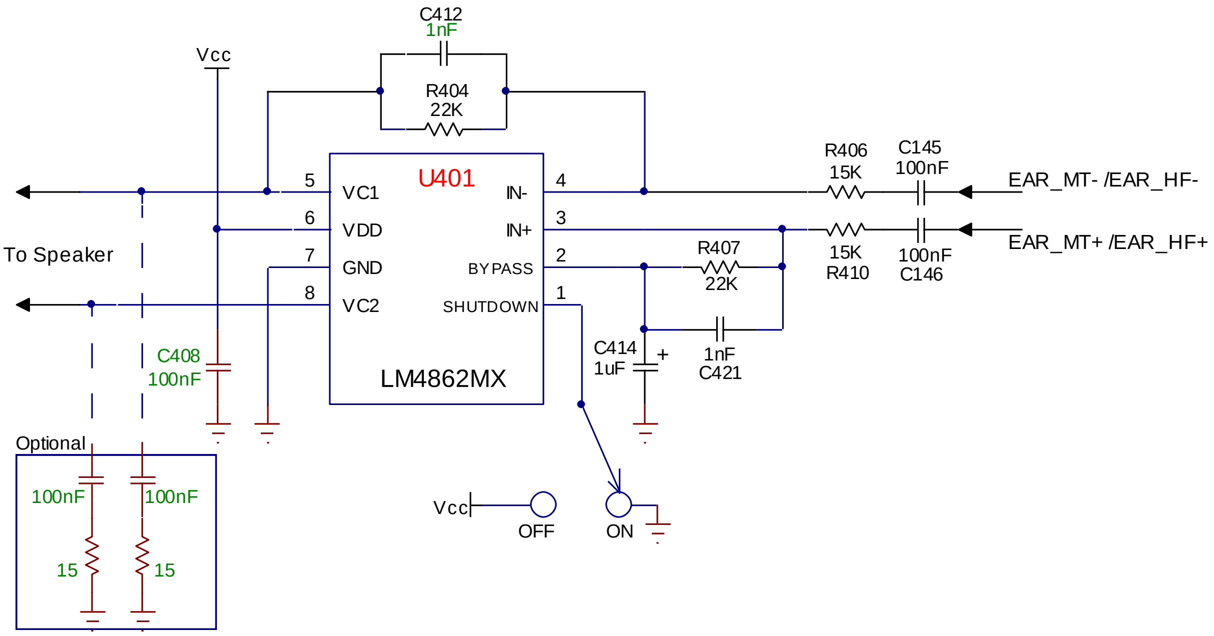

Figure 27. Example of **0.7W** Audio Power Amplifier

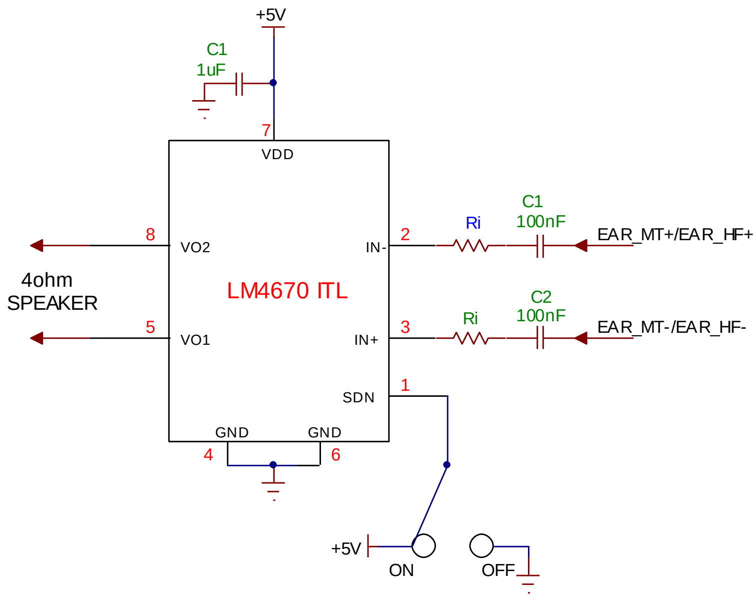

Figure 28. Example of **3W** Audio Power Amplifier

10.5.2.2. Practical suggestions

As mentioned before, you must use an external audio power amplifier to drive the speakerphones or car kit equipments.

The design of such amplifier shall comply with the following guidelines:

- The input to the external amplifier could be taken whether from the “***Ear_MT***” audio path or “***Ear_HF***”
- The amplifier shall have a gain of *(30÷40)times/(29÷32)dB* to provide the desired output power of 5-10W
- If the amplifier has a fixed gain, then its output can be adjusted to the desired value by reducing the input signal by ***AT+CLVL*** volume command of the M2M modules
- the amplifier will have a mute control to achieve the following improvements: when not in conversation the background noise is eliminated and power is saved
- the amplifier and the *modules* power supply voltages should be decoupled as much as possible, by either keeping separate wires and placing bypass capacitors of adequate value close to the amplifier power supply input pads
- the biasing voltage of the amplifier shall be stabilized with a low ESR capacitor (e.g. Tantalum one) of adequate value.

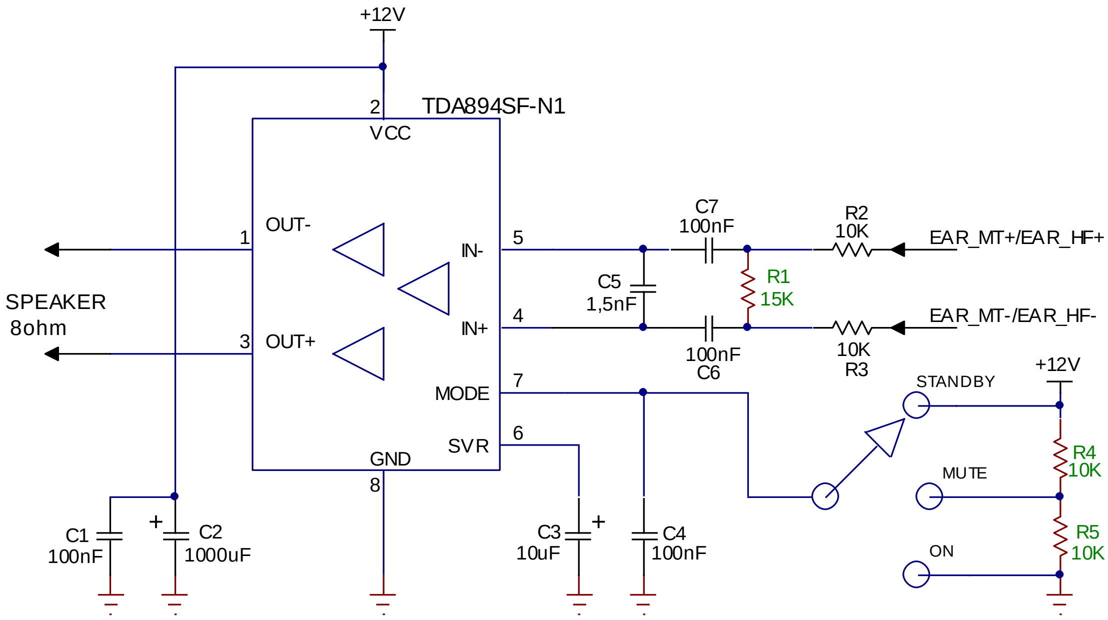

Figure 29. Example of **6W** Audio Power Amplifier

10.6. Audio Interface EAR Gain Parameters

By dedicated AT commands described at paragraph 12.7, the gains of the individual output lines can be adjusted separately so that a maximal degree of flexibility is achieved.

The gain is normalized to ***0dBFS*** that means $3.7V_{pp}$ /differential and $1,85V_{pp}$ /single-ended.

10.6.1. Receive Block Diagrams

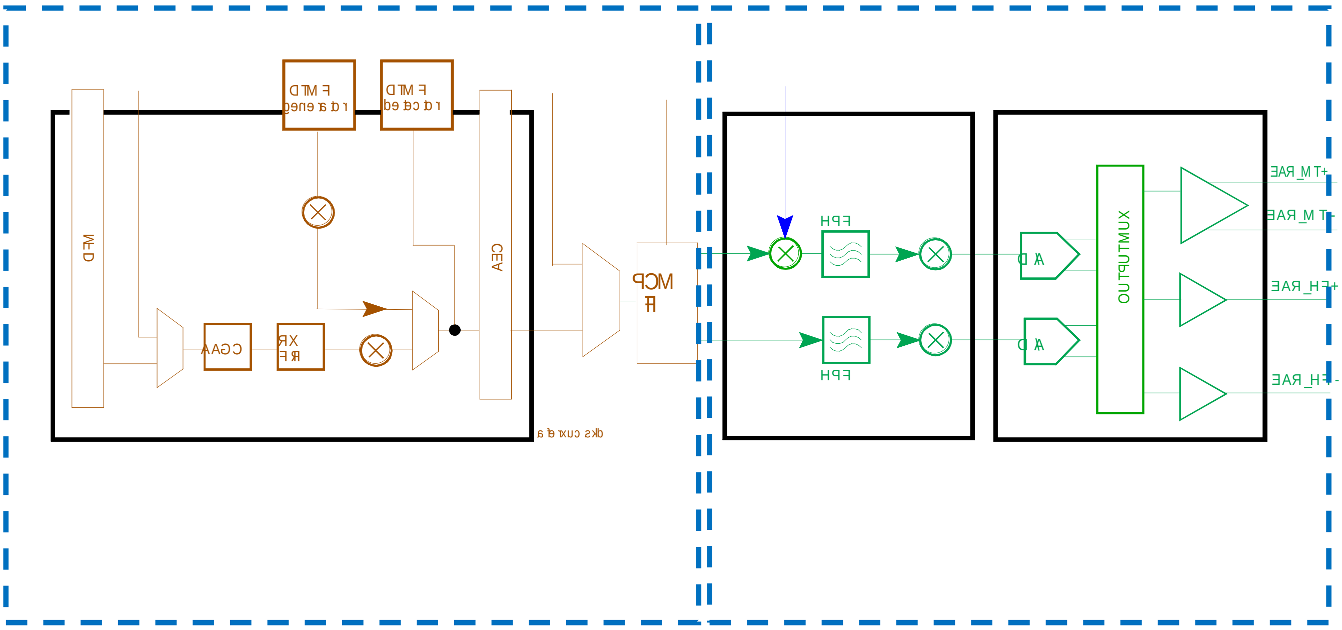

Figure 30. xE910 Families RX section diagram

11. The DTMF detection problem

The standard detection algorithms are not able to recognize the DTMF components sent through a GSM voice channel by the application of the customer.

11.1. DTMF signal characteristics

The Dual Tone Multi Frequency (DTMF) signal is composed by two frequencies as reported in the following table:

	High Group		
Low Group	1209 Hz	1336 Hz	1477 Hz
697 Hz	1	2	3
770 Hz	4	5	6
852 Hz	7	8	9
941 Hz	*	0	#

Table 7. DTMF frequencies

A decoder will give the correspondent digit after detecting the two carrier frequencies according to the table. In order to detect and distinguish the pair of frequencies sent, the common algorithms require usually the total power level of unwanted frequencies to be at least 20dB below the lowest frequency signal with a signal to noise ratio greater than 23dB.

11.2. DTMF generation

11.2.1. First scenario

Responding to the command ***AT+VTS***, the module sends a command to the network infrastructure to generate on the other audio party the correspondent ***DTMF*** signal.
The DTMF tone duration can be controlled partially by the module since it sends a "*start playing tone*" request and a "*stop playing tone*" request and these can be specified by the application controlling the mobile, except from time shifts introduced by the network.
The network infrastructure generates this tone perfectly aligned with specifications requirement, without introducing problem during recognition.

11.2.2. Second scenario

The DTMF signal is generated by a separated source, typically a landline (corded) phone, and sent to the input lines of the module *(Uplink path*). The frequencies couples, sent on the voice channel, are digitized, encoded and sent by the digital transmission system.
In the receiving device the signal would be reconstructed, but since the digital transmission of the voice channel is compressed and optimized for voice, this reconstruction depends on the kind of voice compression used for the transmission, and generally will not perfectly match the original signal.
There are four main types of compression for the voice channel and only the *Full Rate* one has no distortion, while the other three offer a different trouble level (*see figures 31-32-33*):

- *Half Rate.* Problems arise because of the *incoming signal* containing the test signal plus other frequencies, with an amplitude up to *–10dBc*;
- *Enhanced Full Rate*. Bigger problems arise in decoding the *incoming signal* ,that contains the test signal plus spurious frequencies added by the voice compression process, whose amplitude could be very high, up to *–10dB$_c$*. Not only, the two useful components vary continuously theirs amplitude.
- *Adaptive Multi Rate.* This is the worst case, because it is a mixed one.
- *Full Rate* .In this case the *incoming signal* is stable and clean, and there is no problem to decode it since it respects the DTMF requirements. *But it is not applicable to limit the voice coding to only Full Rate , because the network decides itself which coding to be used.*

Figure 31. DTMF 5 dialing with *Full Rate* compression type.

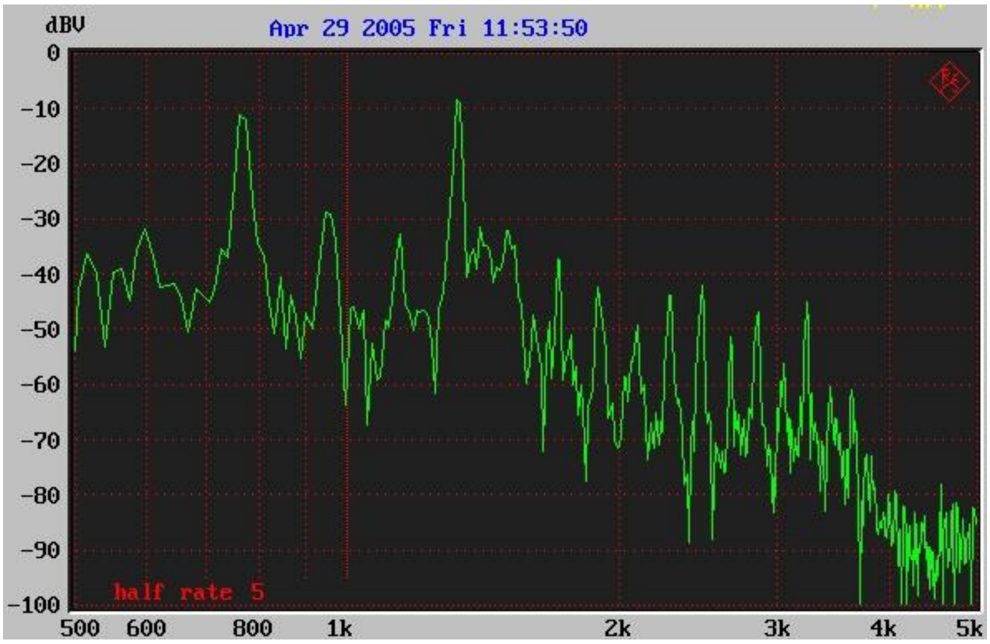

Figure 32. DTMF 5 dialing with *Half Rate* compression type.

Figure 33. DTMF 5 dialing with *Enhanced Full Rate* compression type.

11.3. Suggestions

After these considerations, we suggest the following:

- do not generate *DTMF* signal externally to send them through the voice channel
- send DTMF signal by the command ***AT+VTS***
- set the *SIDETONE OFF* during *DTMF* signal sending

If you need to implement the second scenario, accepting the distortion due to voice channel compression, don't forget that the maximum input lines level is:

1,03Vpp/365mV$_{rms}$ *@ MicGain=0dB*

In fact, higher DTMF signal levels cause saturation of the ***Audio Interface*** amplifier stages, with further unwanted harmonic components generation.

12. Echo cancellation

12.1. Generality

HandsFree systems are equipment that can be used without hands, that is without limiting the movement of the user during a call. They are necessary in several environments:

- in a car's dashboard, while driving
- at the office, during audio or video conferences
- in an elevator, during emergencies
- in open spaces, at the entrance of the parking facilities

These systems are mainly disturbed by the acoustic echo signal that originates from the sound propagation between the loudspeaker and the microphone of a GSM mobile station, sent back to the far user with a significant reduction of the quality.
To overcome this phenomenon, an Acoustic Echo Controller (called ***AEC*** in the rest of the document) is implemented in the Firmware of the Baseband chip.

12.2. Definition

- *AEC* is voice-operated device used for the purpose of eliminating acoustic echoes and protecting the communication from howling due to acoustic feedback from loudspeaker to microphone.
- *Near-end*: anything related to the local user. E.g.: the *Near-end speaker* is the speaker of the *HandsFree system* in a car
- *Far-end*: anything related to the external user. E.g.: the *Far-end speaker* is the speaker of the *PSTN* user.

12.3. Theory

Acoustic coupling between loudspeaker and microphone is an important and potentially negative feature in phones. A mobile phone will transmit via the microphone the direct signal coming from the near-end speaker, the signal coming from the loudspeaker and noise.
Acoustic echo is formed when the sound emitted by a *HandsFree loudspeaker* gets reflected from the walls, ceilings, floor, furniture, people, etc. back to the *HandsFree microphone*. Sound pressure level decreases with each reflection.

12.3.1. AEC Scenario

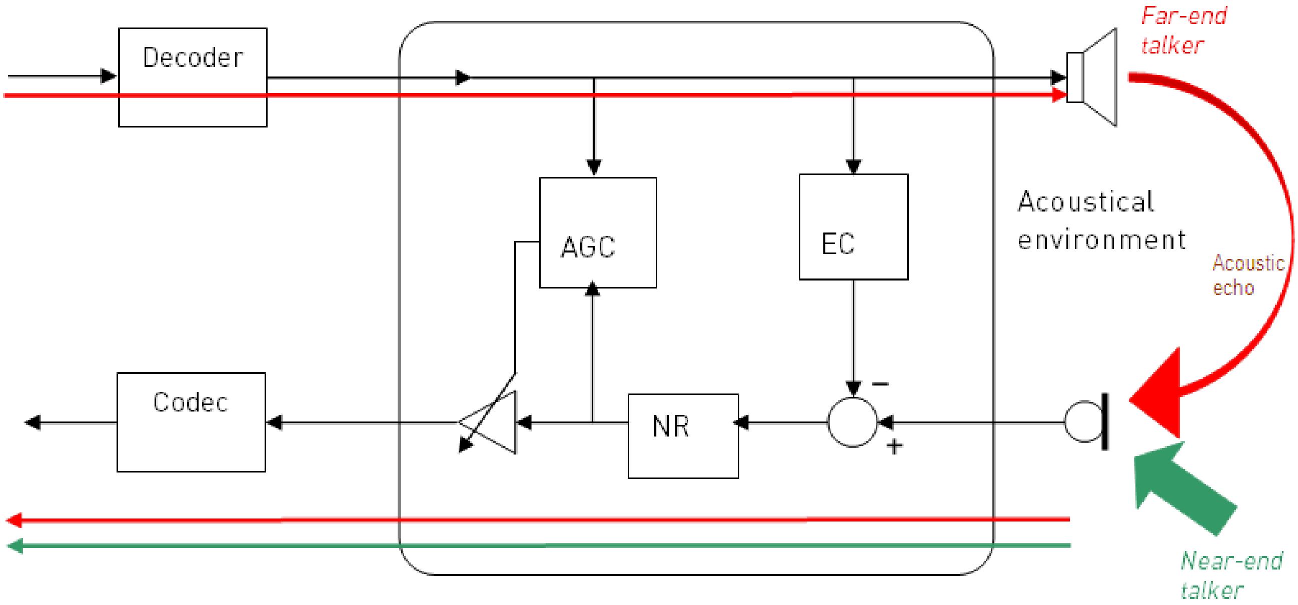

Figure 34. Acoustic Echo scenario

The ***AEC*** implementation consists of 3 modules:

- a scalable, completely time domain based, Block-NLMS (normalized least mean square) algorithm for Echo cancellation ***EC***
- an automatic gain control ***AGC*** (for the transmission path only)
- an additional Noise Reduction ***NR*** (for the transmission path only)

All algorithms, ***EC***, ***AGC*** and ***NR*** are able to operate independently of each other, although both ***EC*** and ***AGC*** are necessary to yield sufficient echo suppression; ***NR*** can work independently.

12.3.2. Audio Interface ECHO block diagram

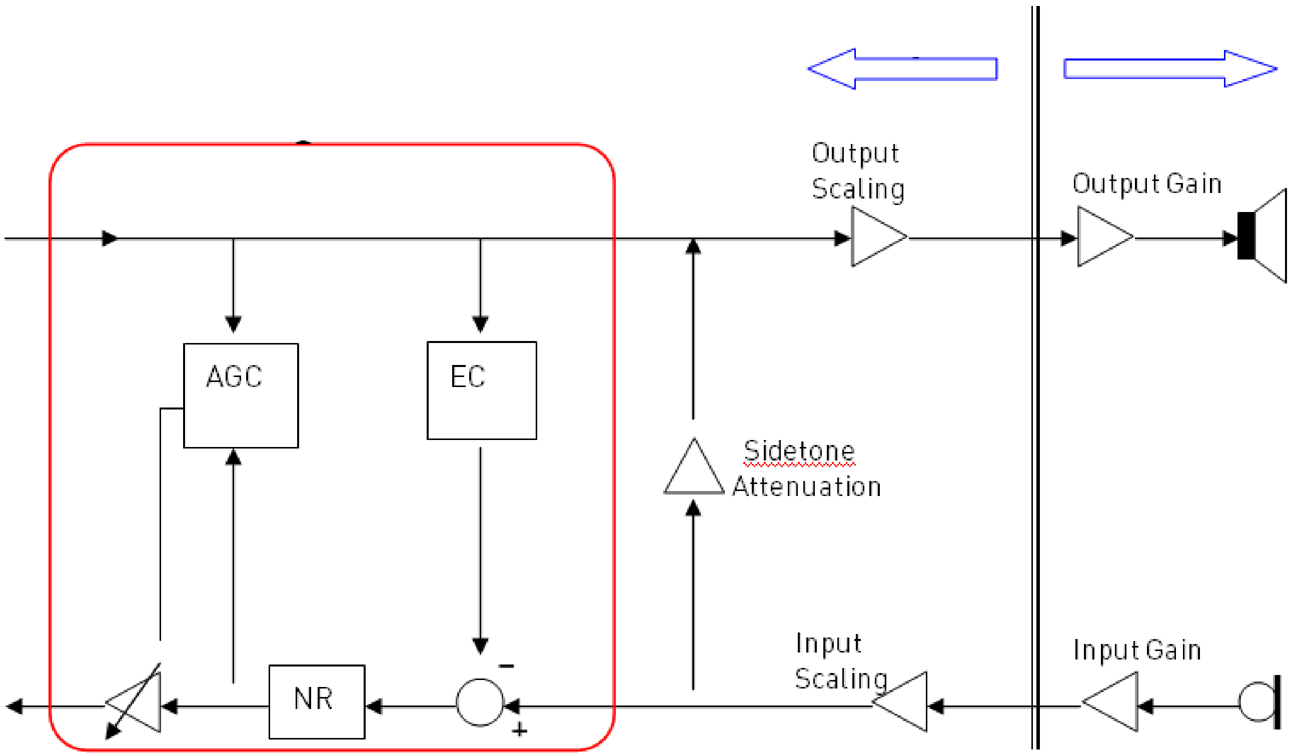

Figure 35. Audio Front End block diagram5

The audio interface consists of the ***AEC*** subsystem, digital scaling and analogue gains.
The ***AEC*** behaviour is configurable by means of 9 parameters, which are settable with a dedicated custom ***AT command*** [1].
Overall Input and Output Gains are to be considered as the cascade of internal amplifiers (*"Output and Input Scaling"* blocks), which gain is settable by ***AT command*** [1], plus any other user implemented external amplifiers (*"Input and Output Gain"* blocks).

NOTICE:

Digital scaling must be used as a way to finely tune gains and correct the chain overall gain.

Due to the original setting of ***HS/MT*** and ***HF*** blocks in the cellular phones (see paragraph 2.2.1), the input digital gain in ***HF*** is 3dB higher than ***HS***, while the output digital gain in ***HF*** is 3dB lower than ***HS***.

12.4. AEC and Audio editable parameters

The following table lists the AEC parameters and Audio parameters that can be modified by AT commands. For further detailed description please refer to the **AT Commands Reference Guide 80419ST10124a and 80399ST1011a** .

12.4.1. EC module

Echo Canceller is based on a FIR filter, and this filter reflects the inverse of the acoustic coupling between speaker and microphone. It can be controlled by the following set of parameters.

- *Adaptation Speed.* It represents the capacity of the algorithm to adapt to the variation of echo characteristics ; higher value means faster adaptation but less accurate convergence and vice versa.
- *FIR Filter Length.* It represents the impulse response length to build the estimated filter, thus it’s the maximum cancelable echo delay.
- *Power Relation RX →TX.* This value is used to tell the algorithm about the real world signal power relation between speaker and microphone and their acoustic coupling.

12.4.2. AGC module

AGC is based on the current signal power relation between RX and TX, and decision is made whether to attenuate or not the TX signal so this block can actually be called “automatic reduction of gain”. The amount of the attenuation is controlled by the set of parameters.

12.4.3. NR module

NR filters background noise introduced by the environment (e.g.: noise of car engine) and the residual noise from the impulse response not removed by the echo canceller, based on the spectral weighting algorithm implemented with sub-bands from 300Hz to 4000Hz, which can be controlled by the following set of parameters:

- *Noise Max Attenuation.* It represents the maximum attenuation that can be introduced by the NR algorithm.
- *Noise Weighting Factor Band 300-500Hz.* A higher value causes better noise reduction but also a higher distortion of the speech signal.
- *Noise Weighting Factor Band 500-4000Hz.*A higher value causes better noise reduction but higher distortion of the speech signal.

12.5. AEC and Audio parameters AT Commands

The following table lists the AT commands available for audio parameters:

AT Commands	Description
AT+CLVL	Loudspeaker Volume Level
AT#SHFEC	HandsFree Echo Canceller
AT#HFMICG	HandsFree Microphone Gain
AT#HSMICG	HandSet Microphone Gain
AT#SHFSD	Set HeadSet Sidetone
AT#SPKMUT	Speaker Mute Control
AT#HFRECG	HandsFree Receiver Gain
AT#HSRECG	HandSet Receiver Gain
AT#PRST	Audio Profile Factory Configuration
AT#PSAV	Audio Profile Configuration Save
AT#PSEL	Audio Profile Selection
AT#PSET	Audio Profile Settiing
AT#SHFNR	HandsFree Noise Reduction
AT#SHSEC	HandSet Echo Canceller
AT#SHSNR	HandSet Noise Reduction
AT#SHSSD	Set HandSet Sidetone

Note that these commands can be applied only under SELINT=2 AT command interface style.

The commands listed below are example, and may not be supported in a specific Module.

For a detailed description of AT commands please refer to the document AT Commands xE910 - Reference Guide 80419ST10124a and 80399ST1011a.

12.6. Practical Suggestions

In this section, an application example will be described, related to a study of echo cancellation in a silent room environment.

The geometry of the system is a very important aspect of *AEC* parameters tuning, so the first step is to decide the spatially setup of the speaker-microphone system and the setting of the gains (using audio parameters and/or external amplifiers), in order to obtain the right speech levels.
The following figure shows the position of the microphone and speakers as suggested by ITU P.340 Recommendation.

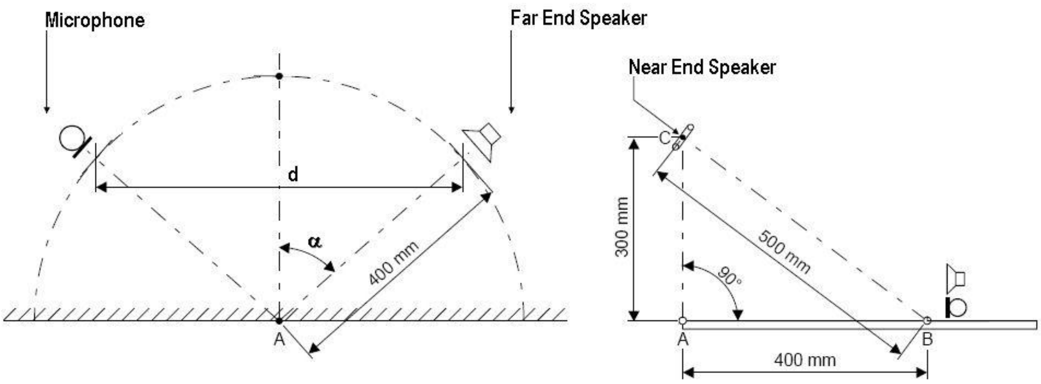

Figure 36. Test setup

We suggest to use an internal microphone gain as low as possible and, if it is needed additional gain to the speaker, to implement it with an external amplifier, using the internal gain only for fine-tuning of the overall downlink gain.

WARNING:

Sidetone is always a harmful effect, so it must be disabled to help echo cancellation action in any HandsFree application.

- Start with the design of the filter length.

The **AEC** has been designed for echo delay *≤50 ms*. For a good cancellation it's enough to take into account only the main echo coming from the near reflections have negligible effects, due to longer delay echoes.
With a distance *d=60cm* in our setup we estimated a maximum echo path *L=10m*, which led to:

$$\mathrm{t} = \frac{L}{v_s} = \frac{10m}{340\frac{m}{s}} = 29{,}4ms \qquad \text{where } v_s = \text{sound velocity}$$

We chose ***40ms*** to have enough margins for the worst cases.

- Consider the level balance of the system.

EC uses the power estimation of the RX signal (**x**) and the power estimation of the TX signal (**y**) for it's coefficient calculation (see fig.34). Because the algorithm does not take in account the gain of the loudspeaker amplifier as well as losses in the air path, something about the real world condition has to be passed to AEC. The parameter to use for this information is the Power Relation RX->TX.

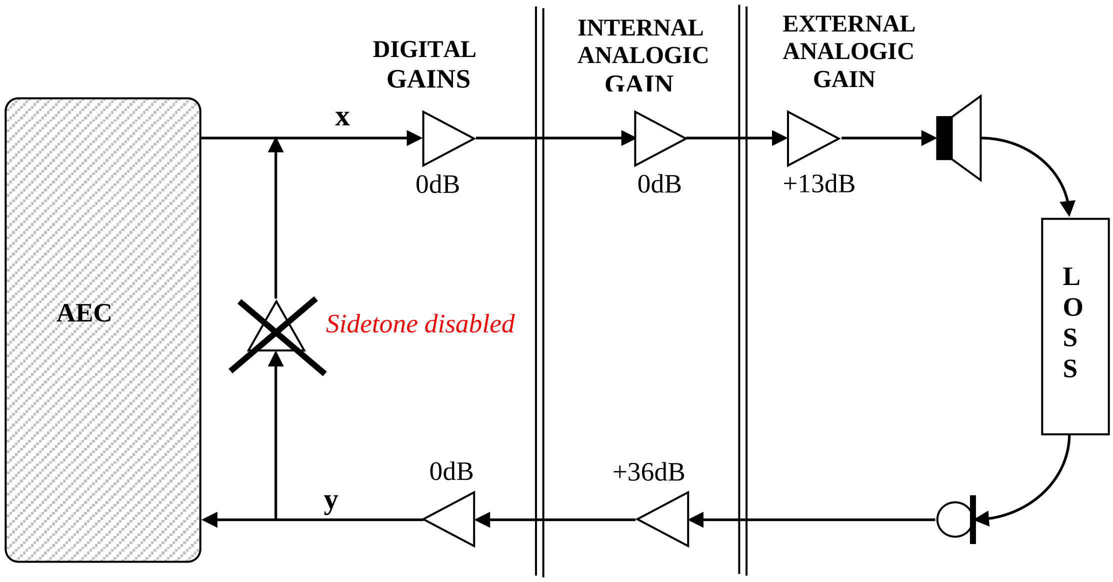

Figure 37. Echo chain

In order to allow the algorithm to compare the power values of ***x*** and ***y***, the following relation has to be respected:

$$y < x + (13 - LOSS + 36)$$

that means :

$$y - (49 - LOSS) < x$$

Estimation of LOSS is a difficult task, because in-air attenuation, speaker and microphone characteristics as well as orientation-depending coupling, lead to a non-repetitive measure of this value. We have estimated this correction factor experimentally spanning the parameter, obtaining a threshold behaviour, which can be seen in following graphic.

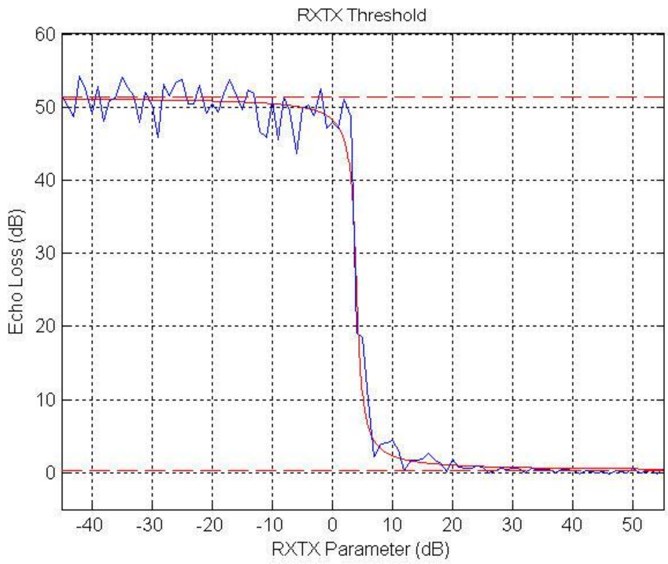

Figure 38. Threshold Echo Loss behaviour

Experimentation in silent room has led to a threshold value of about 5dB.

12.6.1. Warnings about RX →TX relation

WARNING:

Note that this is a threshold value that can guarantee a good behavior of AEC for that configuration; it is a rule of thumb to keep some dB below this value

WARNING:

Care has to be taken, because RXTX works as a threshold value for the AGC and so, in presence of the only NEAR-END speaker, it can be attenuated if speech level is too low.

12.6.2. Suggestions

With configurations that differ from this setup, which means different value for amplifications and/or different spatially position of the transducers, parameter has to be changed following the rules:

- Higher gains has to be compensated lowering the *RX→TX* parameter by same value
- Lower gains has to be compensated increasing the *RX→TX* parameter by same value

- Lower LOSS (because of reduction of the distance between speaker and microphone, or modifying the relative orientation of the two elements) has to be compensated like the case of higher gain.
- Vice versa if the LOSS increases.

When you have adjusted the ***RX→TX*** value taking into account either the *EC* matter, either the minimum level for *NEAR END* speaker, then next step is to set *Max AGC* attenuation, dependent on application and increasing with strength of echo (due to spatial position or big gains) .
Additional attenuation is useful because it permits to the user to add a fixed contribution to the final value of gain calculated by ***AGC***.
We choose 6dB for *additional attenuation*, 0dB for *minimum attenuation* and 12dB for *maximum attenuation*, because these values are a good compromise between reducing echo and maintaining double-talk quality.

On the other hand it is possible to reproduce a half-duplex behaviour setting:

Maximum attenuation=90dB
Additional attenuation=90dB

so if *AGC* kicks due to *RX→TX threshold* (*maximum effect at -90dB*), it will be at *maximum attenuation*.

Intermediate solutions are very closely related to particular application.

Regarding ***Noise Reduction***, it's important to say that its behaviour and tuning are based on subjective tests.
For *maximum attenuation* typical useful values are between 6dB and 18dB, because bigger values determine very poor speech quality.
When tuning the weights parameters it is important to start with the lower ones and then increase values, keeping a small difference between the two (*at least making them equal*).

13. Telit Evaluation kit

Telit supplies the ***EVK2*** to assist the designer during his developing project phase to develop his own applications based on present and future GSM/GPRS/WCDMA Telit modules.
The ***EVK2*** provides a fully functional solution for a complete data/phone application, and is formed by a CS1139B motherboard plus several dedicated *Telit modules Interface Boards* with RF antenna connectors, as shown in the figure 39.

1


Figure 39. *EVK2* (below) with and *GE863-PY* interface (upon).

For further details about ***EVK2*** and its use in designing audio solutions, please refer to the **Telit EVK2 User Guide (1vv0300704)**.

