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Signal Conditioning and Acquisition: General Principles and Applications in the Audio Field

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Abstract: In order to enjoy, store and analyze physical events they must be transduced into electrical signals and, eventually, converted into digital form for processing on computers. This is a standard procedure in every civil and industrial applications: biomedicine, audio, astrophysics, automotive, telecommunications, etc. Methods for acquiring signals and data from the physical world have benefited greatly from technological progress. The latest electronic components greatly improve performance in terms of speed, power consumption, and physical size requirements. Although there are some general guidelines, this process of data acquisition often requires special characteristics specific to the field of application: sensor types, signal conditioning, interfacing, and so forth. Our seminar first describes the general principles of signals and data acquisition. It then focuses on discussion of the analog and digital methods and circuits for the audio field. Regarding this field of application, we will also remark on some psychoacoustic paradigm, some issues regarding audio recording/reproduction and other factors that often address specific solutions adopted for the target systems (telecoms, high-end audio, consumer electronics, etc.).

Key words: analog/digital filters, op-amp, signal conditioning, AD-DA converter.

1. Foreword and general considerations

With some adaptations, this paper follows the DSP Application Day 2010 webinar. We will cover components and procedures for acquisition and conditioning of signals representing events in the physical world, especially in the audio field. Our description will be unavoidably general, due to the sheer vastness of the topic, but we have tried to represent the evolution of audio field technology in a compact way.

We base this study on almost five decades of work carried out by Graziano Bertini at CSCE-CNR, later at IEI-CNR and finally in the Signal & Image Lab (SILab) at ISTI-CNR in Pisa. The most recent studies (including this one) have been co-authored with Massimo Magrini, musician and IT designer and Simone Bianchi, DSP designer and owner of the TangerineTech brand.

The webinar topic is a useful complement to the basic knowledge [1][2] of IT and for electronics people using hardware and software tools to process audio on PC, such as multimedia developers for web, digital libraries and so forth. Media convergence in these fields surely involves audio in a very important way. Some of the topics explained here may stimulate curiosity and interest in a more detailed study.

1.1 Previous ISTI activities in audio field signal processing

In 2003 IEI and CNUCE Institutes in the CNR Research Area in Pisa (figure 1) merged into ISTI, Istituto di Scienza e Tecnologia dell'Informazione "A. Faedo" (named for the famous mathematician).

The ISTI Laboratory devoted to image and signal processing is called SILab, headed by Ovidio Salvetti. In the past the systems developed in our laboratory were mostly destined for other public organizations, mainly for research purposes. Later, SILab started collaborations with private companies as well, such as FIAT-Auto Spa (a project regarding active noise control in the automotive field) and ItalTel Spa (telecommunication applications).

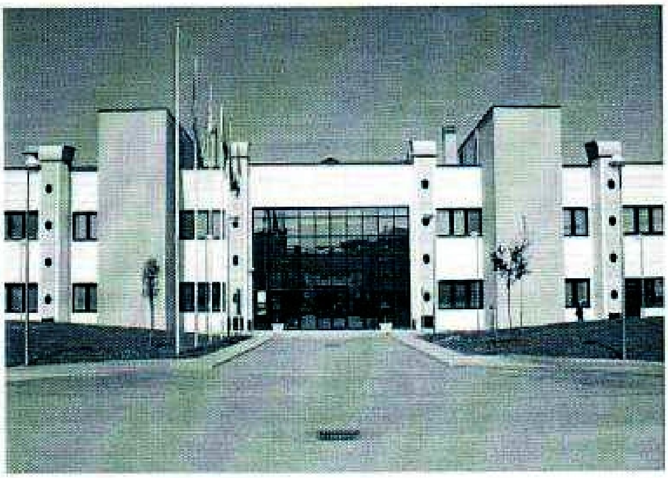

Figure 1. CNR Research area in Pisa.

The development of several DSP-based systems has been carried out in collaboration with various external companies, such as Leonardo Spa (noise control, first DSP boards), SEED Srl (audio processing, experimentation of first Sigma-Delta converters etc.), Audio Devices Snc (audio board tests), Sielcotech Spa for DSP-based underwater modems.

Currently, the SILab has a collaboration with TangerineTech (headed by Simone Bianchi), mostly regarding the design of advanced computer-based systems for high-end musical reproduction.

1.2 From discrete components to mini audio systems based on personal computers.

A brief look at some devices developed at IEI-CNR since the 1960s (figure 2) when transistors became widely available (with all their families npn, pnp, etc). In those years, many projects in the biomedical field were launched at IEI, creating some analog-digital machines with discrete components for multichannel acquisition and pre-processing of physiological signals (optical nerve response, patients' EEG/ECG). Collected data were processed afterward on the general-purpose calculators of the day.

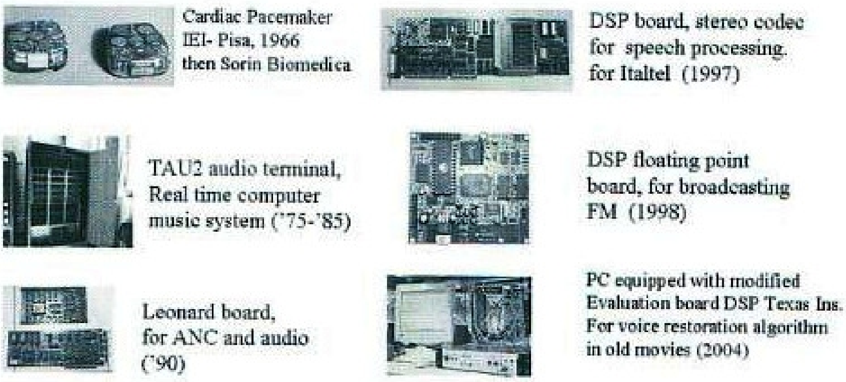

Figure 2. Some systems realized by our laboratory.

A brilliant result was obtained with the design of a self-synchronizing cardiac pacemaker built with low-consumption transistors, resistors and capacitors. The device, first realized in a body-worn version (1966), afterwards in an implantable format, was patented by CNR (F. Denoth, L. Donato) and then taken up by Sorin, a biomedical company in Turin (1967). Early equipment for audio signals synthesis was produced using transistors and integrated circuits, the TAU2 (1975), a big terminal connected to a time-sharing IBM 360 computerr. TAU2 was built entirely in our labs and operated for 10 years for real-time computer music, performing numerous demos and concerts in Italy through remote connections provided by SIP network (today's Telecom Italia) and high-fidelity transmission RAI (Italian Radio and Television).

As the first DSP processors entered the market, board prototypes were designed to enable musical signal synthesis, but results were disappointing owing to the heavy analogue filtering necessary for the A/D and D/A converters of the time, working with a Sampling Frequency of 32 kHz, so noise performance was very poor. Availability of 1-bit serial and then Sigma-Delta converters has vastly improved audio quality, allowing us to fully exploit the DSP chip possibilities.

In fact IEI developed various models of fixed-point DSP boards provided with serial 14-bit mono converters for applications in the field of active acoustic noise control [3][4][5], on behalf of Leonardo S.p.A. company and then with the SEED S.r.L. company leading to the development of many other systems including:

- DSP board with 16 bit mono codec for speech designed for Italtel S.p.A. (1997);
- DSP floating point board for FM audio processing (1998)

More recently a PC test & development workstation for degraded old movie voice soundtrack restoration have been realized in our SILab. The box hardware is based on a Texas Instruments floating point DSP Evaluation Board and other circuits of our own using op-amps for signal conditioning (2005) [4].

2. Seminar summary

This presentation cannot replace proper electronics courses, but helps emphasize key knowledge useful to all audio designers and operators. In particular, we should point out that analog technology knowledge is still a key curriculum requirement for those designing digital systems that interact directly or indirectly with the physical world, since real-world signals are almost always analog in nature. The main topics covered by the webinar and described in the paper are:

- General considerations on electronic systems
- Analog chain and its main characteristics
- Signal conditioning and AD/DA converters
- Acquisition techniques evolution: oversampling and Sigma-Delta converters
- Examples of signal conditioning
- Hard disk recording
- Recent designs by ISTI-CNR in the audio field.

The paper begins by referring to the general benefits and opportunities offered by electronics in general with both analog and digital technology, especially in audio. The next section describes some well-known equipment and also specific analog electronics and their main functions. We then proceed with the description of the digital chain, providing details to illustrate the various components for the conversion from analog to digital and also the opposite passage, from digital to analog. Particular attention is devoted to examining AD-DA sigma-delta converters, the most useful and ubiquitous device for 15 years (but little known in their operation). We also mention digital protocols used to link audio devices and some methods of conditioning signals using external standard or specialized cards connected to a PC. Finally, we refer to some recent achievements and projects in which cited devices and technologies are applied in general and in our lab.

3. General basic considerations

We can perceive physical events in two different ways.

i) The first and obvious one is a direct perception: the physical world event is evaluated only with the use of human senses or with the aid of simple mechanical instruments. For example, listening to a heartbeat using a stethoscope, measuring temperature with a thermometer, measuring pressure with a barometer, listening to acoustic instruments in a concert hall without amplification, or evaluating the disturbing noise level coming from an engine.

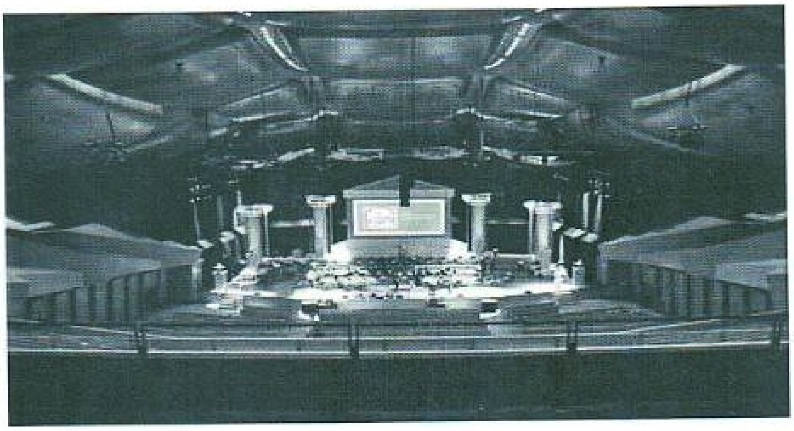

Figure 3. Auditorium in Rome.

Since ancient times musical events have been enjoyed in specially crafted venues (e.g. Roman amphitheaters). Even today, whenever possible, acoustic music is played and enjoyed without any electronic reinforcement (figure 3). This is obviously true for symphonic and classical music.

ii) The other way to evaluate a physical event is by using electronic devices. The use of electronics has offered many advantages and much better results and objectives in many situations.

Just think of the audio industry, where the operation of recording for the storage and use over time and at a distant place. In the case of a concert being played in a bad environment, electronic systems can provide a better presentation of the event in real time i.e., inserting delays on the outputs and strengthening power in the sounds at the same time and same place where the performance takes place. Obviously, the chain of transmission of information consists of other components, of which we provide a brief description

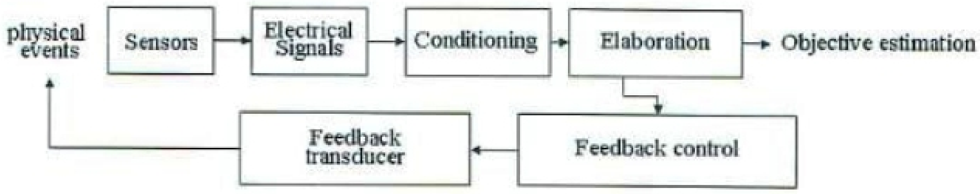

Figure 4. A typical block diagram for signal processing and control.

Sensors transform physical parameters into equivalent electrical signals. These signals are then managed and conditioned depending on the purpose of the whole system. Electronic signal transfer can therefore solve some intrinsic acoustic problems and make possible the long-distance transmission/archiving of live events.

Signal conditioning is the processing of electrical signals required for transferring the signals to the final stages (visual or acoustic representation) with the best precision and signal-to-noise ratio. This kind of processing is commonly performed by amplifiers and filters made with circuits built using operational amplifiers and discrete components, explained in the following.

The use of electronics is very useful in the example cited in Point 1) allowing an objective measurement of physiological parameters, environmental data and so on. In industrial applications, where in addition to the acquisition chain that improves the direct measurements, the electronic system may include a feedback loop that ensures the stability of process control for changes in working conditions.

The final processing blocks and even some conditioning are currently being implemented with digital technology and only in specific applications with the analog one; in this regard we shall see an example in the audio field in the following paragraphs.

3.1 Psychoacoustic elements.

In order to design electronic systems for audio processing it is necessary to know how hearing works, as well as the relations between physical measures and their related properties once transformed into electric signals.

Sound is basically a waveform of air pressure variation, and it is transformed into an electric signals using some kind of sensors (microphones etc.). We know that fundamental sound features (timbre, loudness, pitch) are strictly related to the signal parameters (spectrum, amplitude, fundamental frequency) [6].

It is also known that the variation of these parameters are linked to perception of the others (when transformed into sound again) in a non-linear way. For example regarding loudness, we need to consider the Fletcher&Munson curves (figure 5).

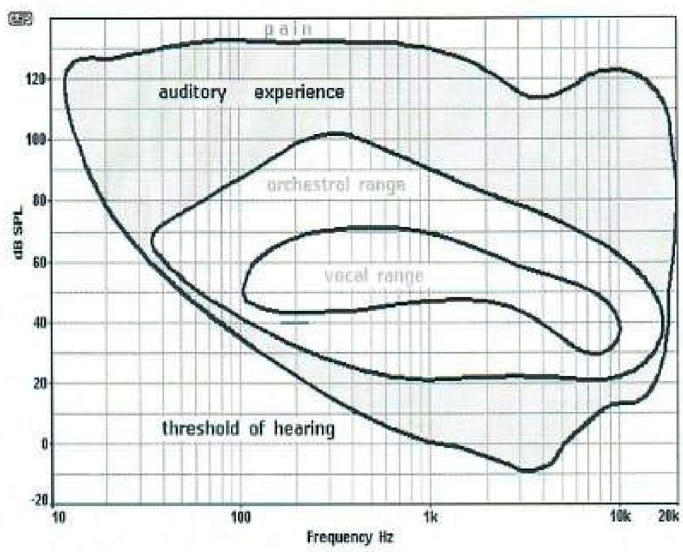

Figure 5. Fletcher & Munson curves with different areas.

For example a consequence of such behavior is the loudness filter, traditionally present in audio preamplifiers, that enables the signal amplitude to increase at high and low frequencies (bass and treble), when volume is set at low levels.

The typical range of values of dynamic and frequency field of the vocal, musical and full audible field is also represented in figure 5. This characterization is important in order to defining the design specifications of the circuits used for their treatment. A basic rule is generally taken into account for choosing the audio bandwidth and dynamic according to the target of the system as follows:

- Musical applications: full range (0-18.000 Hz, 20-120dB)
- Telephony: vocal range (300-3000 Hz, 30-50dB)
- Hearing aids: extended vocal range (200-5000 Hz, 30-100dB)
- Measuring noise instruments: full range (0-20.000 Hz, 0-130dB)

In the telephony field, the reduced bandwidth and dynamics allow the use of low-cost 8bit-8kHz audio data converters that are less expensive than their high quality audio (music) sophisticated counterparts, running at 44.1 kHz – 16 bits or more.

Sometimes signal conditioning, if not well implemented, can affects the quality of the perceived signals. Obviously, some kind of alterations may be desired (i.e. musical effects), but sometimes they can lead to artifacts that should be avoided. For examples saturation (distortion), when not used as an effect, has to be avoided. As we know, musical instruments are characterized by the spectrum of their sound, and how a peak waveform envelope evolves in time. If we exceed in the use of dynamic compression this instrument's "fingerprint" may be altered and, for example, the typical "punch" of a percussive sound may be lost (figure 6). For a more detailed discussion of these arguments we suggest reading the book *Mastering Audio*, by Bob Katz [7].

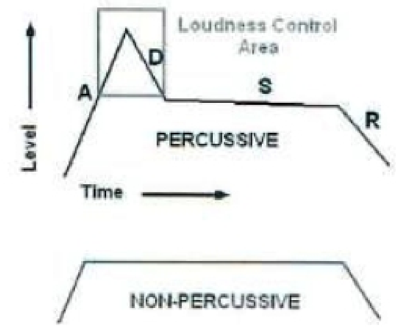

Figure 6. Heavy envelope modification by dynamic range compression.

Some other, more complex sound processing aspects involving psychoacoustics are forward masking, backward masking, frequency masking, etc. In some contexts these aspects may lead to artifacts, but in other situations (i.e., mp3 compression) they can be usefully exploited.

4. Audio analog electronics

Besides the direct use of amplification and broadcasting of audio events in any location, the use of analog technology, thanks to the development of recording and reproduction, has enabled tremendous progress in human activities with respect to cultural artistic, social, and industrial aspects. Vinyl records, magnetic tapes and cassettes led to the replication and dissemination of audio events in different locations and with a lag in time.

An example of a analog chain is the PA system (Public Address) for the reinforcement of live performances figure 7). This is the well-known system shown in the block diagram of the Figure where the main steps of the manipulation of the electrical signal are to be played: the microphones convert the sound waves (event) into an electrical signal, the mixer provides the impedance matching, amplification, checks the right levels.

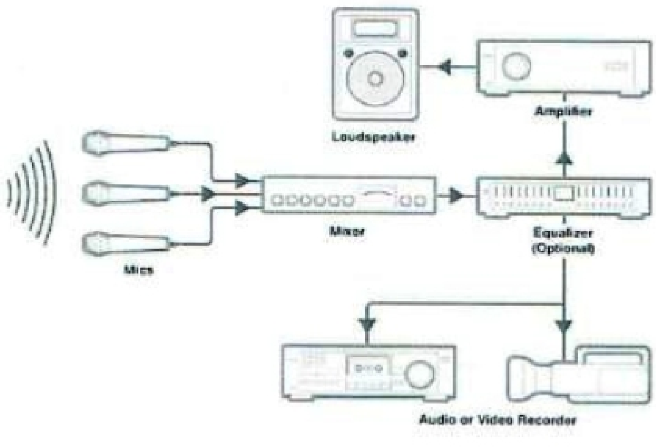

Figure 7. A PA System.

Then the signal is sent to an equalizer which has the task of controlling the frequency response of the system and distributing the signal to the next and final stages.

An analog chain can be basically represented by the schematic structure of the figure 8:

Figure 8. Analog chain for signal treatment .

In this diagram *sensors* convert a physical quantity (event) to be measured into an electrical signal.

Examples of sensors: audio microphone (many types), pickup for guitar (piezoelectric, electromagnetic), ultrasonic probe, photoresistors, phototransistors, photo/infrared diodes etc.

Interfaces or front-end between the sensors and the other stages may include the following functions: power supply (phantom microphone), isolation, impedance matching, other signal adjustment (balanced/unbalanced), etc.

The block labeled *conditioning* here realizes signal modifications as: amplification, filtering/equalization, dynamic range modification (compression, expansion), etc.

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Of course, digital technology with its high processing ability, besides improving final sophisticated functions, also allows the realization of some of the above functions taking place in the analog components. This is well known thanks to their diffusion in the consumer market (pedal guitar digital effects, small storage devices, media streaming and more).

4.1 Conditioning.

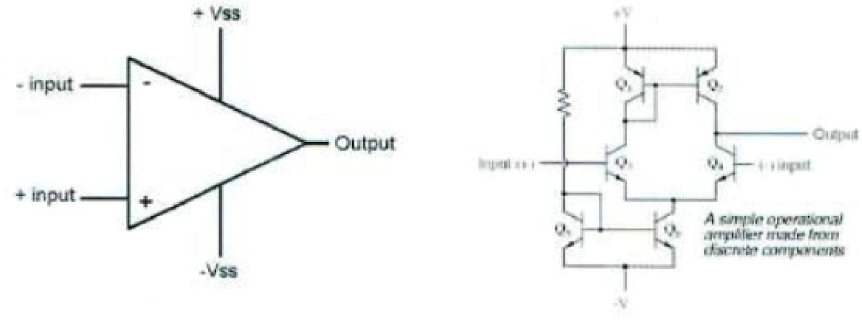
The correct choice and use of sensors and interfaces of the analog chain requires considerable practical experience and we suggest that the reader consult other specialized bibliography, reviews and product manuals; here we offer only a few elements of some functions, and components of conditioning circuits.

Amplification is needed to adapt the current and voltage characteristics of signals to those needed by the following stages and can be effected by a wide range of devices such as transistors, vacuum tubes, and mosfets, employed in various circuit architectures.

Filtering is used to enhance frequency ranges to be treated, or to reduce unwanted ones. This can be obtained using full analog solid state or vacuum tube filters or, more recently, by means of integrated, switched-capacitor filters. Other conditioning operations are dynamic range processing (compression and expansion of the signal amplitude) to improve the signal-to-noise ratio of the whole chain.

Final processing at the end of the analog chain can simply consist of power amplification or in other more complex operations (i.e., power modulation in radio transmitters). The latter case will be described later in this section.

Let us present some details about a very common component in conditioning circuitry, the operational amplifier (op-amp with its schematic symbol in figure 9a). It is based on a differential amplifier, ideally having infinite open-loop gain and input impedance (figure 9b). Adding a few discrete components and changing the feedback configurations, it is possible to obtain several analog functions, such as sum, difference, integral, filtering, etc.



(a) (b)

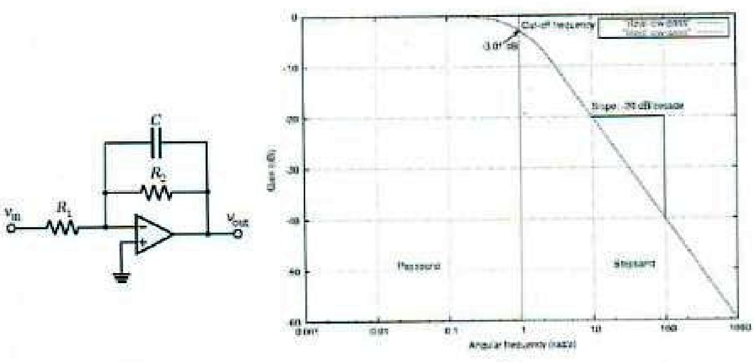
Figure 9. Operational amplifier: a) symbol, b) simplified internal schematic

Even whjen the operational amplifier is available as an integrated circuit, sometimes a discrete component version is preferred, mostly for better signal-to-noise ratio. Monolithic op-amps are marvels of technology, but when performance is critical, they cannot match a discrete op-amp. A discrete op-amp costs more and is larger than a monolithic op-amp, but it offers superior performance in many ways, due to S/N ratio, linearity, stability, and heat dissipation. This is because every different component is fabricated on a manufacturing line that is fully optimized for that specific part. Therefore, each component is the very best on the market. As a drawback, a discrete op-amp is larger and more expensive than a monolithic op-amp.

Figure 10. A discrete component op-amp: the Hardy 990 "building block" device.

For example, the 990 type from John Hardy Co. supports many critical audio applications such as microphone, phono and tape-head pre-amps, as well as amplification functions (figure 10).

Obviously, skilled designers can design their own amplifiers based on transistors as well on thermionic tubes. A simple configuration of a low pass active filter with an operational amplifier is shown in figure 11a with its response (figure 11b).



(a) (b)

Figure 11. Low pass filter with op-amp and it's response.

Circuits like this are very common in audio systems, mostly for limiting the audio band to the range of interest and improving stability and dynamics. They are very common before the A/D converter too, although modern converters have built-in low pass filters. The cutoff frequency depends on the component values: F= 1/(2PI*R2*C), with a center band gain equal to −R2/R1. This filter is quite simple (first order), so its transition band slope is only 6 db/oct (20dB/decade) and does not fit well in certain audio applications (figure 11b).

Filters of the second and third order (2 or 3 poles) are obtained and even single-stage Filters with more orders (8 or more poles) are obtained by connecting in cascade multi-stage op-amp. There is a limitation to the number of poles, as it exalts the noise due to various effects (thermal, etc.) mainly introduced from the first stage. There are many other kinds of filters, more complicated but with better performance (in term of ripple, stop band noise, transition band slope, etc.), such as Bessel, Butterworth, Chebyshev, Equiripple, and Sallen-Key: vendors supply user guides and sw tools in order to facilitate the design for specific requirements.

For special applications, specific circuits have been studied. For example, the circuitry figure 12 shows a differential amplifier that has input interfaces provided with high stability and buffering capacity on both input branches, with the task of increasing the input impedance of the block. This configuration addresses high-level applications, i.e. for measurement instrumentations.

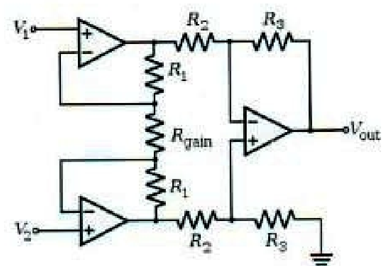


Figure 12. Differential amplifier.

Gain may be regulated varying the Rgain resistor. This is the expression of the relation between output and input voltages: Vout = (1+2R1/Rgain)(R3/R2)(V2-V1).

Another typical conditioning circuit is shown in figure 13: it contains the basic functions of an audio dynamic range compressor by analogue technology.

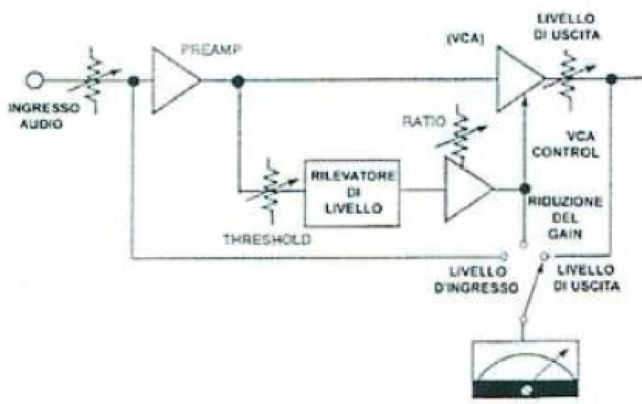
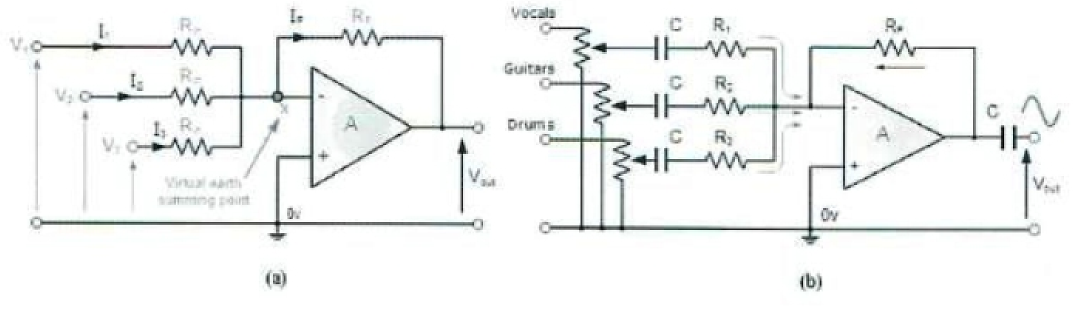


Figure 13. Analog dynamic compressor typical schematic.

The input signal is measured and its level is compared to a preset reference threshold. When the signal level exceeds the threshold, a voltage controlled amplifier gain (VCA), which controls the input signal pathway, is reduced. This causes the dynamic swing of the output signal to decrease. A voltage meter is switched between the input and output signal in order to evaluate relative amplitude.

4.2 A brief comment of final analog processing examples

Once the signals are properly brought to the desired level in amplitude and frequency towards the final output of the analog chain they can be subjected to appropriate analog treatment for various applications. For the sake of thoroughness, we give some examples.



(a) (b)

Figure 14 a) Adder, b) Audio mixer

The summing amplifier is an electronic circuit which uses an operational amplifier to add two or more signals (figure 14a); the output voltage Vout is the sum of each Vin controlled by the ratio of Rf/Rin. With the insertion of some blocking capacitors (figure 14b) we obtain a simple audio mixer for adding or mixing together individual waveforms (i.e. musical signals) from different source channels (vocals, instruments, etc.) before sending them combined to an audio amplifier. This summing circuit is used in many audio devices and can be considered an analogue basic signal processor.

Another example is Amplitude Modulation , which has been used since the beginning of the radio broadcast industry (early 20th century) up to now .

The information-carrying signal (i.e., audio) is multiplied by a high-frequency signal (carrier), apt to be broadcast by radio transmissions, obtaining the AM signal used by radio stations (figure 15a). Both examples are nothing other than mathematical operations on signals performed within the domain of continuous time by means of the transfer characteristics of the devices. These are not operations performed as explicit calculations, as in digital systems.

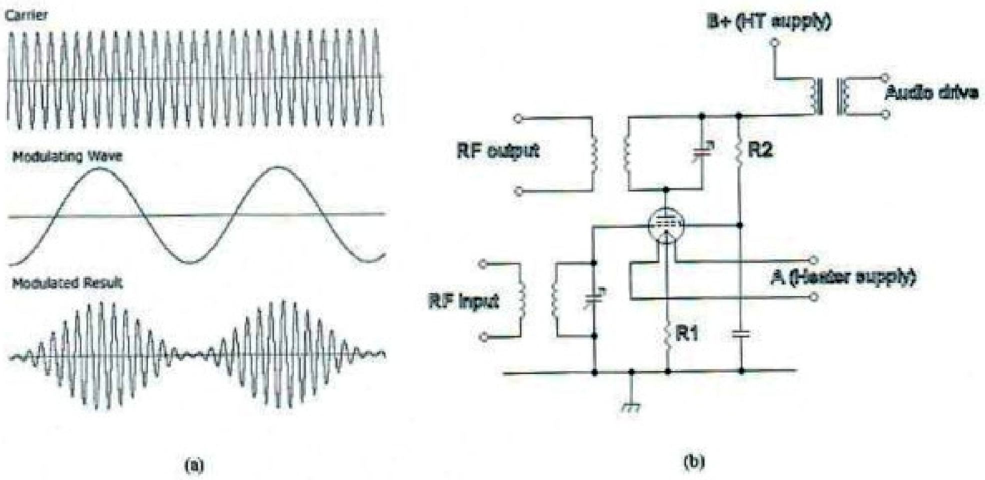


Figure 15. Amplitude modulation: a) waveforms, b) vacuum tube based modulator

In practice, the AM modulation can be achieved either through transistors or even with valve (vacuum tube) technology, as shown in figure 15b with an AM modulator anode coupled to the transformer. The vacuum tube is a tetrode type, whose anode and screen grid power is modulated through the coupling transformer denoted as Audio drive. This means that the input signal high-frequency (RF input) are amplitude modulated and the output is then transferred to the RF output to be amplified in power and sent to the antenna.

4.3 Design tools

There are many SW tools to support the design of electronic circuits employing discrete components and op amps, which allow you to simulate how the circuit should behave, at least in theory. Among the most popular tools running on a PC we mention Pspice, Labview, Microcap, and Filter Wiz Pro.

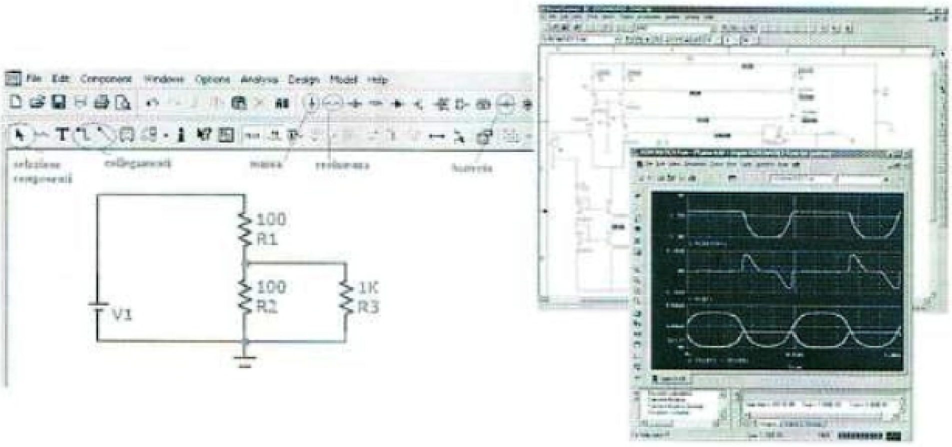


Figure 16. Examples of an electronic design tool.

In figure 16 we see how to design the components of a circuit by choosing the types and values from the palette. Once the project is complete, you can "place" some oscilloscope probes at certain test points and visualize the voltage levels or the waveforms if the input is a signal generator. Obviously, effective practice results may differ more or less from those simulated, for example due to differences in the real value from those of the plate, or many other factors, such as parasitic losses in the printed board tracks and other causes not always identifiable a priori.

4.4 Analog chain problems

Among the various causes due to analog components that produce signal distortion we will mention at least three. One drawback is due to the non-linearity of the active device transfer functions (vacuum tubes, transistors, op-amps, etc.) that give origin to an alteration of the output signal harmonic content. Figure 17a shows the spectral analysis of the output of an amplifier whose input is monochromatic (sinusoidal): the spectral lines exceeding the original frequency are the harmonic distortion's components (multiple frequencies with respect to the input one). Since the information in analogue systems is carried not encoded but directly by the signal characteristics, so distortion tends to change the signal's information content. This alteration is defined by the abbreviation THD (total harmonic distortion) and measured as the ratio of the total power of unwanted harmonics vs. the power of the fundamental test tone.

Another problem exists in all circuit types, both analogue and digital, that is the electronic noise caused by random current and voltage variations due to thermal charge carrier movements in all circuits components: figure 17b shows a noise-disturbed signal. For example, noise could be reduced by lowering the circuits' working temperature: this method is used only in research centers of physics experiments. For usual applications electronic noise reduction is obtained by specific circuit design solutions.

The other type of noise can be induced by high-frequency electromagnetic radiation coming from an external or internal source: however, a well-made routing design and shielding solution drastically reduces this inconvenience (using multilayer boards, etc.)

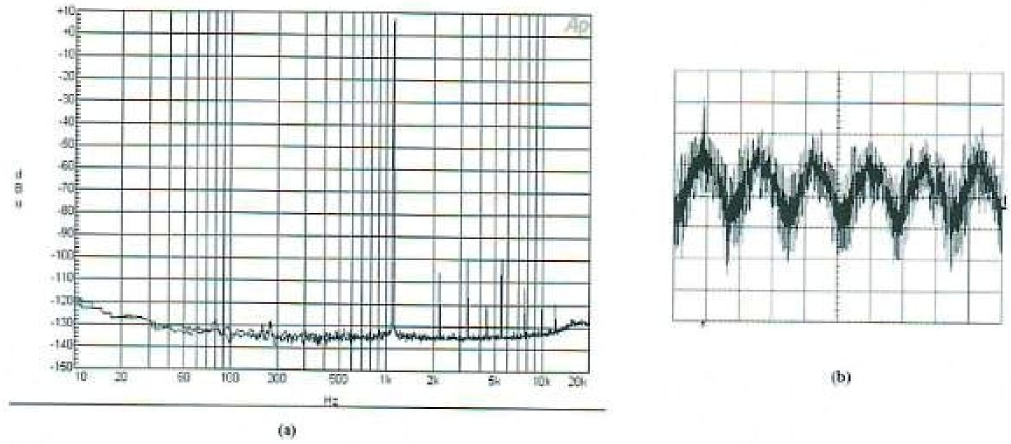


Figure 17. Analog chain problems: a) harmonic distortion, b) noise

In all cases of analog signal treatment, information is disturbed by noise with the signal that carries it. In the case of digital signal generally associating signal discretization with signal symbolic coding, noise plays a critical role only in the phases of A/D and D/A conversions, while during signal processing in symbolic form (i.e., PCM) it is sufficient that the noise levels do not impair the transmission and decoding of the symbols representing the signal itself. Some main analog chain characteristics are listed below:

Pros

- Simple
- Controllable
- Copies the event without coding it
- Maximum resource economy

Cons:

- Only simple processing available
- Every passage in the chain worsens signal characteristics
- Impossible to separate the event from its representation
- Signal storage is always at a loss
- Recording and replay always imply mechanical components

5. Digital acquisition conditioning and data converters: main components of the digital chain

Let us now consider the components needed to make an acquisition and processing chain by digital audio technology, introduced in the 1980s and now established in all areas of human activity. The analog signals from microphones or other sensors after they have been adapted by front-end, are sent to an analog-to-digital converter (A/D): this component provides the output with bit strings representing numbers expressed in binary form, proportional to the signal analog values (figure 18). This makes it possible to achieve the required processing (filtering, effects, modulation, etc.) using algorithms implemented in software on processors (general purpose or specific digital signal processor-DSP) and exploit all the possibilities of such systems [8][9][10]. Among the numerous advantages of the digital chain are noise reduction, stability, versatility and reproducibility, for all processing and storage of digital data. At the output of any digital signal processing is always necessary to perform the reverse translation, that is, digital data must regain the analog information. This function is accomplished by a digital-to-analog converter (D/A).

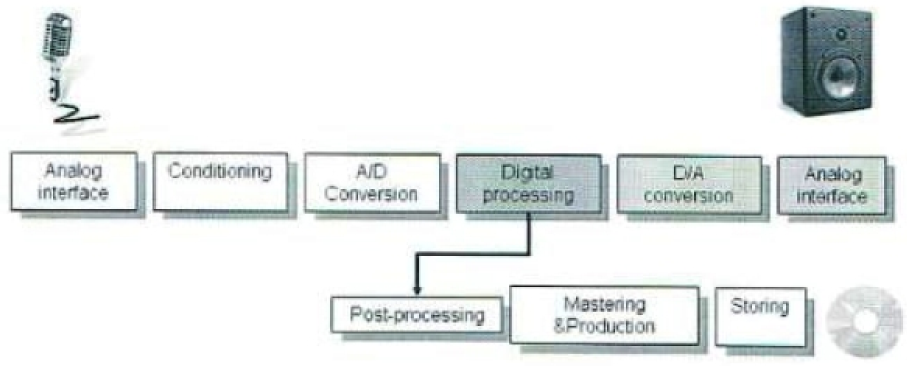


Figure 18. Digital audio chain block diagram.

However this approach to the treatment of audio signals requires the use of analog parts in order to perform the functions well in the acquisition section (input of A/D) and vice versa for filtering the analog signal from digital to analog. This is necessary so that digital processing operates correctly on signals free from noise and distortion.

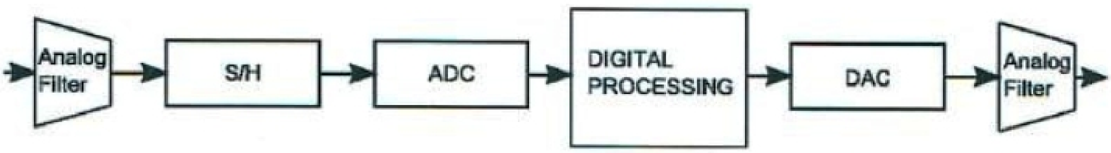


Figure 19. AD/DA in details .

A digital chain originates when the analog signal representing the event (typically time-continuous) is transformed into a discrete signal (in both time and amplitude domains). Time discontinuity is obtained by using a Sample and Hold device (figure 19), which samples the original signal at a predetermined instant of time, holding its amplitude constant up to the following sampling time.

The following device, the A/D converter, quantizes the signal amplitude, with a number of amplitude levels fixed by the device resolution (related to its number of bits). The output of the A/D converter is made up of a stream of samples, representing a code (PCM) with specific voltage levels.

The digitized signal can then be processed using numerical techniques (i.e. explicit calculations, not strictly dependent on the physical properties of the devices carrying the signal) implemented in either hardware or software on numerical processors, exploiting this way the countless possibilities offered by Information Technology. The processed samples can be stored to be read back later, or sent to the D/A converter that produces an analog signal again in real time. However the D/A output suffers from discontinuities, which will be removed from a specific low-pass analog filter.

In addition to the number of output bits, very important parameters for both converters are the signal-to-noise ratio and the conversion time (i.e., the maximum sampling rate). Later we shall see some types of these important components and their integration into the audio chain and requirements to be satisfied by fields of employment.

5.1 AD and DA conversion details in the frequency domain

The Sample and Hold device allows transformation of the original input signal (band limited according to the Nyquist theorem) into a time-discontinuous signal (that is to say, varying only at sampling times, neglecting its evolution between them).

However, here the signal can assume any value in the amplitude domain. Amplitude discretization is performed by the A/D conversion device next in the chain.

Sampling introduces discontinuities in the time domain that means multiply its spectral content in the frequency domain.

If signal bandwidth has been correctly limited to under half of the sampling rate (B<1/2fs), the spectral replicas that form will not overlap themselves or the original signal spectrum.

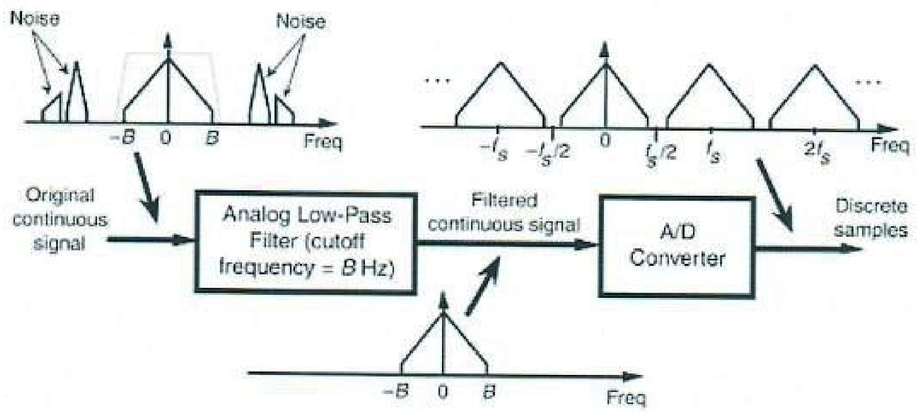


Figure 20. Avoiding aliasing with well designed lowpass filter.

The possible overlapping phenomenon is called aliasing and must be avoided. The low-pass filter prior to the conversion device is called anti-alias filter It must have a high slope in correspondence to the transition band and a high attenuation in the stop band, in order to remove the undesired frequencies while leaving the desired signals (figure 21).

The simplest first-order low pass filter has a poor 6dB/oct (figure 11a) while an 8-pole low pass type Sallen-Key filter is a more complex circuit for a 48dB/oct transition band slope (figure 21).

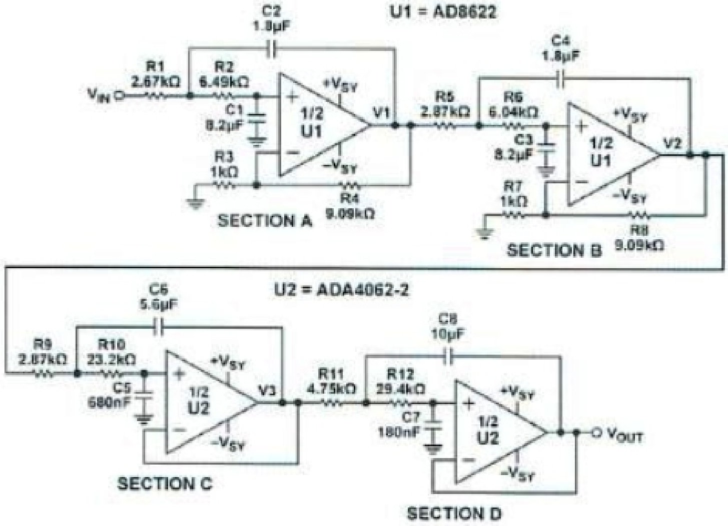


Figure 21. Sallen-Key low pass filter.

An efficient system for performing analog filtering is the switched capacitor filters where the filtering curves are obtained using only a series of capacitors switched with very high frequencies. These circuit solutions are very compact and are implemented in a single chip without using external resistors. However, since there is no optimal solution for audio quality due to induced noise, we do not go into detail here. A/D converters often incorporate D/A units; we will examine the classic D/A converter used in audio since before the Compact Disc and popularized by the standard introduced by Philips and Sony in 1980.

The signal lines corresponding to every bit are used to switch calibrated resistors (figure 22), changing the value of the current flowing in the output line. A current-voltage converter transforms this output into a voltage available for the following low-pass filter which removes signal discontinuities from the system output.

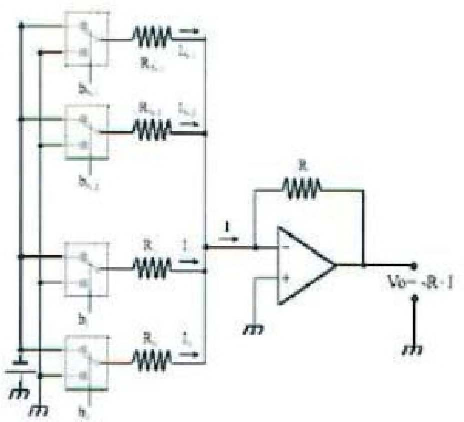


Figure 22. Weighted resistors DA converter.

A relevant type of converter is the Multi-bit SAR Converter (figure 23). This type of A/D converter uses a D/A converter and a comparator to obtain the bit representation of the input analog signal.

The sampling timing signal simultaneously activates the Sample and Hold unit and a controller that starts to count sending to the D/A its counter value. D/A output level is then compared with the Sample and Hold signal until the best approximation of the signal itself is reached. Now the bit representation of the input signal is taken to the output of the conversion unit.

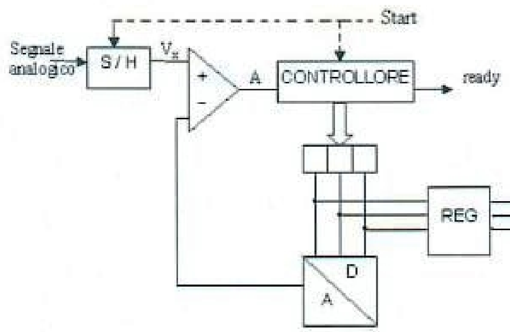


Figure 23. AD converter with SAR technique.

This conversion method requires a number of clock cycles equal to the allowed number of levels, that is to say 2n where n is the converter bit number. Other methods allow more rapid conversion in only n steps, adding a little hw complexity on the controller; one of these techniques is the successive approximation register (SAR). However, due to their operation mode, this kind of A/D converter is available for audio application up to 100 kHz sampling rate and 16-18 bit depth.

To obtain greater speeds (frequency response) 'Flash' A/D converters can be used (figure 24).

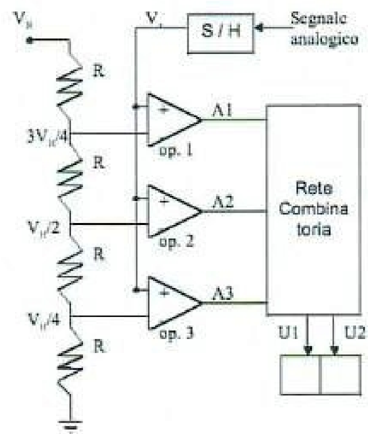


Figure 24. Flash AD converter.

Flash A/D converters are employed for low-resolution/very high speed of conversion. The analog sampled signal is compared with the thresholds (properly scaled) of a comparator battery. According to the number of comparators that result in a positive output, a logic network obtains the bit representation of the input sampled signal. These converters can reach extremely fast sampling rates (Ghz) with low resolutions (6-8 bits).

5.2 AD-DA conversion problems

The first problem is the approximation in the A/D conversion operation that introduces a systematic error in the digital chain (figure 25).

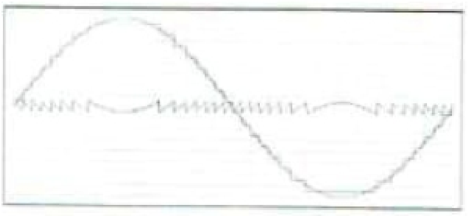


Figure 25. Digitizing error.

The red (stepped) line represents the digitized signal, the green (continuous) one the original input signal, the blue (along the horizontal axis) line their difference: the last one is the quantization error. Depending on the type of application (telephony, voicemail, music etc.) we need to use the highest possible sampling frequency and converter bits number, in order to render this type of error negligible.

The other difficulty is the need to obtain a steep transition between passband and stopband of the input and output analog filters of the digital chain. In fact, the anti-aliasing filter before the A/D converter must reject all frequencies above ½ the sampling rate with an attenuation of 80-90 dB or more. The same requirements are valid for the reconstruction filters after the D/A converter; all the spectral replicas (which are adjacent to the useful spectrum) must be removed from the output signal. Early CD Audio players (44.1 kHz, 16 bits) suffered from the difficult design compromises between sharp attenuation (12-14 poles low-pass filter) and its low noise/high stability requirements.

Even electromagnetic interference between digital audio data bus (wideband, discontinuous fast signals) and analog audio lines are a serious inconvenience in digital audio data circuit design.

6. Advanced audio data conversion techniques: Oversampling and Sigma-Delta converters

The problem of multiple lines carrying the digital audio data signal was first resolved with serial mode data transfer, in which all bits of a sample are sequentially transferred between digital audio devices (converters, data interfaces, etc.) on a single line, while two other lines carry the synchronization for sample start (FrameSync) and bit clock (CLK): for the audio stereo it needs six wires, three for input and three for output

- Oversampling is a family of techniques that allows running A/D and D/A audio data converters at frequencies that are much higher than the actual sampling frequency of the digital signal stream. This way, by using Digital Signal Processing (DSP) techniques, the analog filter requirements can be relaxed, allowing the use of simple low-order low-pass filters which do not present stability problems.

- Sigma-Delta technique has been used in audio applications since the 1990s and allows the use of advanced DSP processes for designing very high-performance audio converters that do not require substantial calibration, enabling audio applications to reach a refinement (in the high performance field) and a quality/price ratio (in the budget field) that were not possible before.

- Sigma-Delta conversion technique. Initially only used in telephony due to the limited performance, it was improved making it indispensable for all audio applications since the 1990s.

The development of these techniques offer several advantages: either A/D and D/A or both integrated in a single chip (in the latter case called codec); the use of a simple RC anti-alias and reconstruction filters; up to 24 converter bits and the extension of sample rate Fs up to 96 kHz and beyond. While it is fairly easy to understand how the serial output mechanism operates with the bits of digital samples, we provide an explanation for the other two techniques.

6.1 Oversampling

Oversampling is a mechanism that allows, starting from a common digital signal, to enable the reconstruction of the analogue version by using simpler analogue filters.

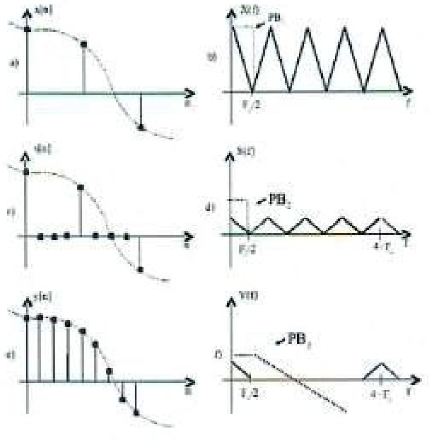


Figure 26. Oversampling.

In figure 26 we see the representation of a sampled signal and its spectral content. The spectral replicas of the original signal are very close and must be removed with a very steep (high order) analogue filter, which is complex and prone to realization difficulties. Digital signal frequency is then increased by inserting zero-valued samples (in this case three) between the real samples. This way the digital stream has reached a bandwidth that is quadruple that of original (sampling frequency is now quadruple).

Those which were originally spectral replicas are now well within the bandwidth of the digital stream and can be removed by using of a high-order digital filter (FIR or IIR) that easily removes them without artifacts. In the time domain this is equivalent to interpolating the added samples in order to align them smoothly to the original ones. The first real spectral replica is now very far (four times the old sampling frequency) and can be removed with a low-order low-pass filter without difficulty: the larger operation frequency of the DA converter is not a problem. Figure 27 shows the block diagram for a classic oversampling solution.

Figure 27. Block diagram of oversampling DA.

Digital filtering operations can be performed by external components or integrated within the converters. Digital filters are essentially made up of delay lines, multipliers and summers. By changing their operation coefficients we can obtain a vast assortment of filtering operations that are extremely difficult to obtain using analog processes.

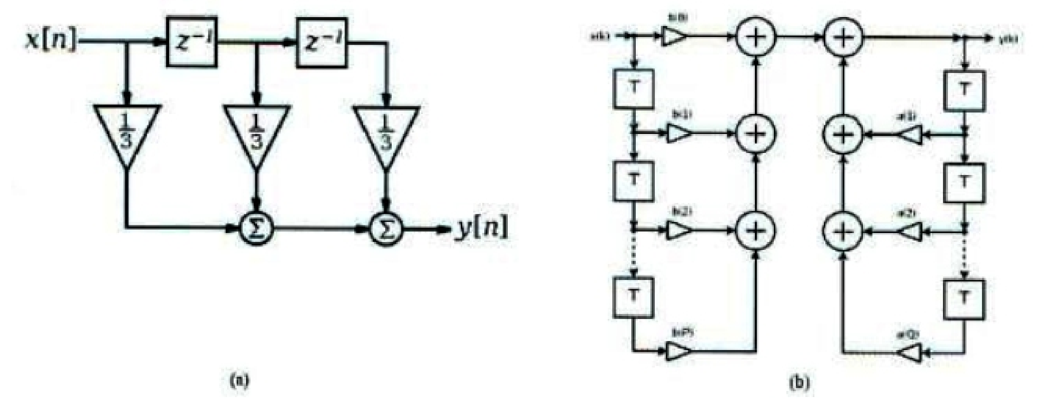


Figure 28 FIR and IIR digital filters structures

A FIR (Finite Impulse Response) filter is characterized by the lack of feedback elements (figure 28a). This kind of filter is computationally expensive, but has stability advantages.

Some converter manufacturers adopt IIR filters (Infinite Impulse Response), characterized by feedback elements (figure 28b). These filters are more efficient in term of processing time than FIRs, but their stability is not always guaranteed.

6.2 Sigma-Delta converters

A Sigma-Delta A/D consists in a Sigma-Delta modulator which produces a 1-bit modulation and a low pass filter (figure 29).

A modulator obtains the Sigma-Delta bit stream starting from the analogue signal in a way similar to a PWM modulator. The modulator switching frequency can be chosen much faster than needed in order to obtain an effective oversampling (ie. a sampling that is much faster than needed for the representation of the input signal).

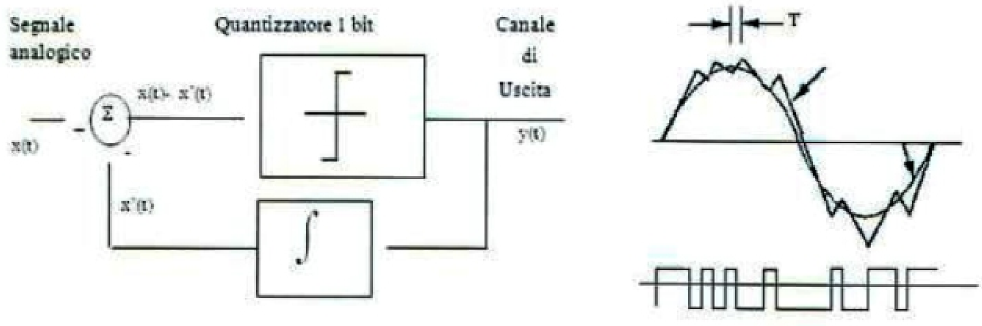


Figure 29. Sigma-Delta AD conversion principle

The Sigma-Delta signal is then sent to a digital low pass filter which yields a multibit (even if serially transferred by a 3-wire interface) PCM output (figure 30).

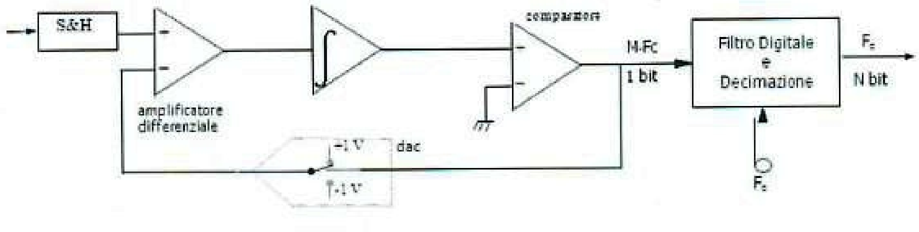


Figure 30. Sigma-Delta with digital low pass filter

In a similar way a D/A will consist of a digital modulator which transforms the multibit PCM signal into a 1-bit Sigma-Delta bit stream and an analogue low pass filter that re-obtains the analogue continuous signal. The actual circuits are rather complicated especially in parts dedicated to noise reduction: the mechanism is known as noise-shaper and is illustrated in figure 31.

With certain configurations of Sigma-Delta modulators we can obtain a non-constant spectral distribution of the noise produced by the conversion operations.

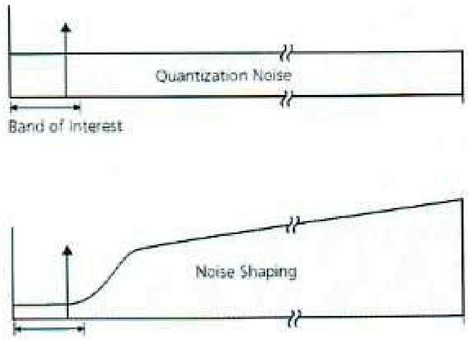


Figure 31. Noise shaping

In particular, noise can be concentrated in the higher Sigma-Delta bit-stream spectrum, so as to be radically attenuated after the output low-pass filter. While this produces a strong increase in performance in the useful band, these devices have a tendency to radiate high frequency noise and need an accurate circuit layout in order to keep these products from affecting the following analogue stages. Starting in the 1990s, Sigma-Delta conversion chains have taken the place of native PCM converters in many high-performance PCM audio applications.

7. Other analog conditioning techniques also used for digital chain input-output

In order to improve noise performance without complicating specific circuit parts, some applications adopt pre-emphasis and dynamic range compression techniques (figure 32).

For example, pre-emphasis filtering boosts the high spectrum range before A/D conversion, in order to be able to use an additional low-pass filter (de-emphasis) after the D/A converter output.

This way the high-frequency noise residuals of the entire conversion chain are proportionately reduced, compared to the useful signal high frequency contents. Signal-to-noise performance of the chain is therefore improved, but maximum input levels at high frequencies are reduced. Wide spectrum signals will distort more easily at high frequencies compared to emphasis-less chains.

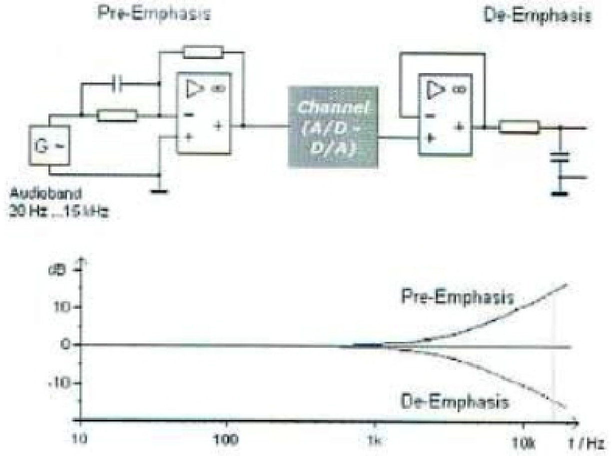


Figure 32 Emphasis and de-emphasis technique.

The dynamic de-emphasis system proposed by Analog Devices in the early 2000s consisted in a low-pass variable filter controlled by a signal detector. When the useful signal level drops below a threshold, variable filter bandwidth is progressively reduced from 35kHz to 1kHz, practically attenuating the channel background noise. This system does not need pre-emphasis, but produces signal modulation artifacts, which are easily perceived.

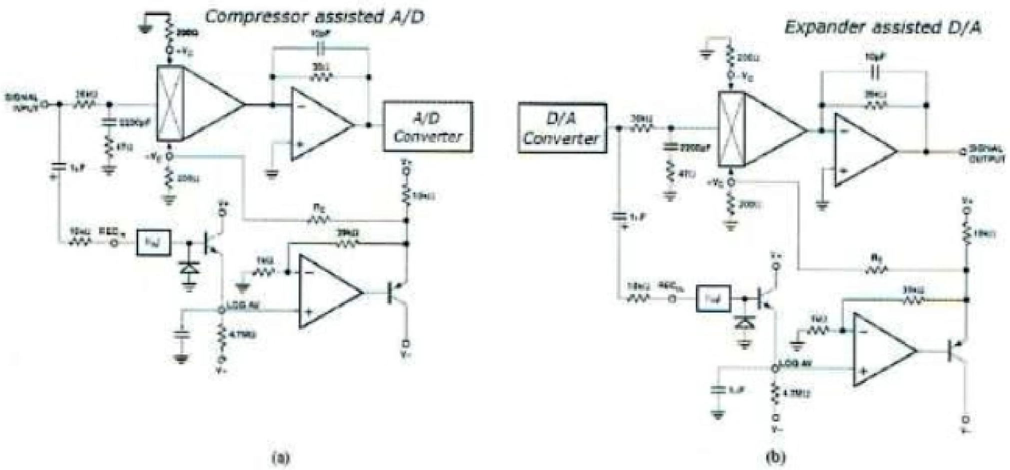


Figure 33. Compression assisted AD and expansion assisted DA.

A technique widely used in the telephony field and also used in other applications is to compress the input signal dynamic range prior to A/D conversion and (optionally) re-expand it after D/A conversion. This allows obtaining a digital stream with a reduced bit number signal whose original dynamic range would require a greater bit depth. The figure shows an implementation of a dynamic range compressor used to supply an A/D converter input. Weak signals are amplified (in order to mask the conversion chain noise) and strong signals are attenuated to avoid conversion device saturation.

The circuits shown in figure 33 implement this technique. The expander characteristic must be exactly specular to that of the compressor in order to re-obtain the original signal without alterations. Since obtaining perfectly specular compression/expansion characteristics is impossible, a certain quantity of distortion added to the original signal is unavoidable. Compressor/expander assisted systems are therefore used when digital channels with a adequate bit depth for the useful signal dynamic range are not available.

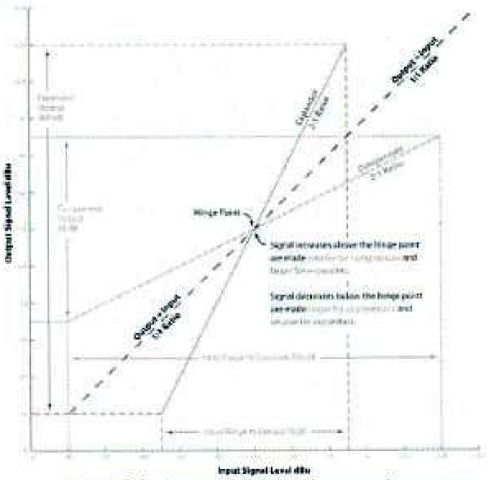


Figure 34. Compression and expansion curves

In figure 34 we compare the expansion/compression characteristics of the respective chain stages. Their composition (in the case of perfect specularity) produces a linear characteristic and thus the original signal. The compression/expansion characteristics illustrated in the figure allow for a reduction in signal dynamic range from 100 dB to 50 dB during the signal transmission, expanding it back to 100 dB after the D/A conversion.

7.1 Some mention on Digital Audio signal routing

Most electronic manufacturers produce daughter boards with AD/DA converters for test and evaluation purposes. Digital audio devices are generally interconnected making use of a fast digital connection consisting of 3 lines (figure 35): Data, Data sync and Frame sync (for discriminating left and right, or more, channels).

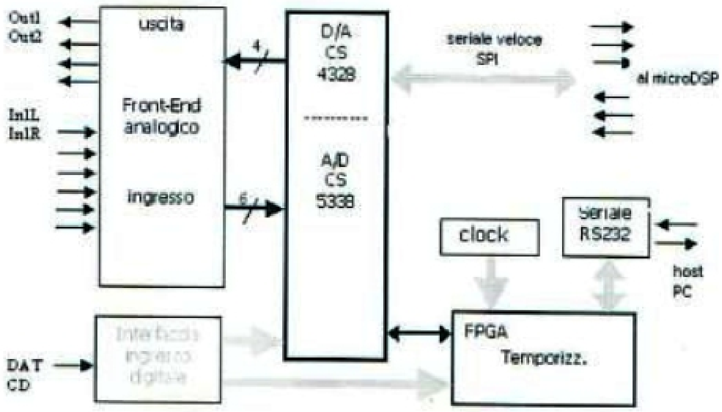


Figure 35 AD-DA connections

Digital audio appliances (CD Players, Set-Top Boxes, Tape Recorders, etc) are generally provided with an optical digital input and/or output. This way it is possible to connect devices (e.g. CD->DAT, or DAT-Audio card) without quality loss.

The most common format is S/PDIF, very similar to AES/EBU protocol.

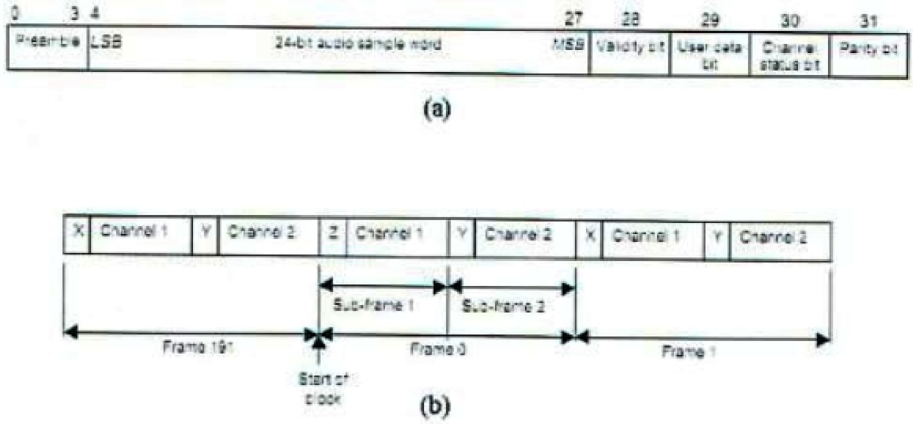


Figure 36. AES-EBU Format

The digital audio standard frequently called AES/EBU (figure 36), officially known as AES3, is used for carrying digital audio signals between devices. It was developed by the Audio Engineering Society (AES) and the European Broadcasting Union (EBU). Several different physical connectors are defined as part of the overall group of standards. A consumer variant of the standard, S/PDIF is also available.

The low-level protocol for data transmission in AES/EBU and S/PDIF is largely identical, and the following discussion applies to S/PDIF as well unless otherwise noted.

AES/EBU was designed primarily to support PCM encoded audio in either DAT format at 48 kHz or CD format at 44.1 kHz.; it allows the data to be run at any rate, and recovers the clock rate by encoding the data using biphase mark code (BMC).

The bit stream consists of 64-bit frames transmitted once per sample time. This is divided into two 32-bit subframes (or channels): Channel 1 (left) and Channel 2 (right). Each subframe consists of 32 time slots used to transmit individual data bits or synchronization information. Twenty-four bits are available for audio data, of which 20 bits are normally used. A total of 192 consecutive frames are grouped into an audio block. Certain status information is transmitted once per audio block. At the default 48 kHz sample rate, there are 250 audio blocks per second. It can use various cabling and connecting systems: balanced, unbalanced, optical, coaxial.

7.2 Final considerations on digital chain performance

A great deal of the processing in the digital chain is performed on special DSP processors or on general-purpose computers. DSP processing. Intensive digital processing is mainly operated by algorithms implemented by specific DSP software; detailed discussion is beyond the scope of this paper, but we list some pros and cons of the entire digital compared to analog chain.

Digital Audio Chain:

Pros:

- Complex processing possible, impossible in the analog chain, is available
- Signal can be transferred without quality loss
- Signal can be stored without quality loss

Cons:

- Complexity
- Demanding in term of processing power
- Signal must be encoded symbolically
- Transmission channel must be better than the analog one, when the same information is transmitted.
- Quantization noise and frequency response limitation are unavoidable (but can be set in such a way that their effect is practically negligible).

8. A case study of hard disk recording and recent ISTI-CNR projects in the Audio field.

In this part of the paper we cite some examples of signal conditioning, both in professional audio applications and in SILab design.

8.1 Hard disk recording

Most of today's recording studios use computer-based systems for recording and processing musical signals. For many years personal computers have featured sound cards (often integrated in the main board), but only recently have they been used extensively in the studio: thanks to the processing power of modern CPUs, all of the processing that was originally performed by analog equipment (EQ, effects) is now done with software.

Typically recording studios use external audio boards, in order to minimize possible interference between digital PC circuits and analog signal sections. Internally these boards use the same type of professional converters (i.e., Sigma Delta) discussed in the previous paragraphs: the connection to the computer uses some other specific digital interfaces (i.e., Firewire) that are faster than AES/EBU. Pro and semi-pro audio devices usually have built-in microphone-level pre-amplifiers, with balanced inputs such as XLR or TRS with balanced line and phantom-powered mike-level inputs (figure 37a). However, in studio applications the direct connection of a microphone to an audio board is not the rule. Despite a perfect recording, it is impossible to obtain an exact copy of the acoustic original, even using the most advanced devices. Recording engineers usually make use of some signal processing and effects to improve the quality of the rendering. Thus, it is more correct to imagine the recording as a 'painting', and not a 'Xerox copy' of the original event.

(a) (b)

Figure 37 a) Echo-digital Audiofire 4, b) Vacuum tube based analog front-end

A recording is a representation of the acoustic event as seen by the recording engineer, who is responsible for the acquisition, processing and (sometimes) mastering of an album. Very often studio engineers use special signal conditioning to give more "feeling" to an acoustic instrumental recording.

The most realistic sound is not necessarily better; we should never forget that the final sound will be listened to by a human, not a measuring instrument. The best recording is the most emotional one, the one that is most realistic to the listener. For this reason audio device manufacturers have proposed various amplifier solutions, each adding its particular character. For example vacuum tubes are often used to "warm sound", an expression used to describe the second harmonic enrichment of the sound spectrum. It is very important to note that although there are many digital and software emulators of these effects, none of them are comparable to the original, produced by an analog one. For this reason, for signal conditioning professional studios will continue to use special pre-amps, including tubes proposed by old well-established as well as emerging companies. As a real-world example, we consider a small company tube preamp designed by Diego Barone (figure 37b). It is used as a front-end for a digitizing board, and was made for a well-known recording studio order.

The harmonic enrichment obtained with this preamplifier is basically a distortion and not a small one, since it is around 1-2% (note that HI-FI systems typically have a distortion of 0.1% or less).

If this setup is used to record a voice, its timbre will be much warmer and more alive. Sound obtained with a direct connection of the microphone to the AD board is colder, with a harsher sensation.

8.2 Case studies of ISTI Signal & Image Lab designs

In the latter part of the paper we cite some examples of signal conditioning currently adopted in our lab for recent audio projects.

- Pandora Project

This project concerns the development of a system for real-time parameter extraction from musical instrument signals in a multimedia application. The microphone is connected to an audio compressor for amplifying and adapting the real-world signals to the input stages of the DSP equipment, in this case a common personal computer without special sound cards.

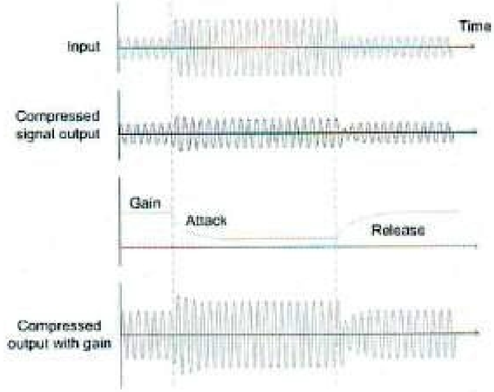
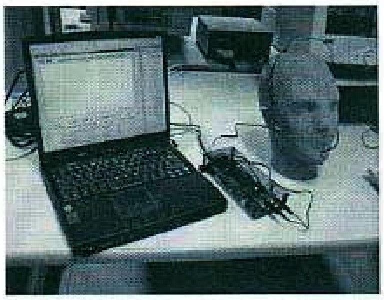


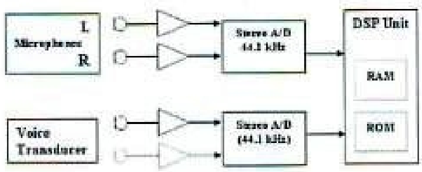
Figure 38. Dynamic compression in Pandora system

The problem is caused by the relatively high SW threshold to be introduced on input signals caused by the poor quality of the PC audio section. By dynamic range compression (figure 38), the low level signal waveform is raised, thus allowing the correct analysis by the DSP algorithm running on the PC.

- *Precision acoustic noise dosimeter*

Current noise dosimeters for daily dose assessment are generally provided with a single microphone. ISTI-CNR worked on a project regarding the design and the realization of a system for accurate noise measurement (figure 39a) provided with a multi-channel input, in particular: two channels for a binaural recording, two other channels used to detect and subtract the wearer's voice contribution, which could alter measurements. The prototype uses a starter kit made by Analog Devices, and all the conditioning circuitry (especially necessary for the piezo-electric transducers) was subsequently added (figure 39b). Among the measurements performed by the instrument are Interaural Time Difference (ITD), vocal emission by throat-mikes, and strip-lines.



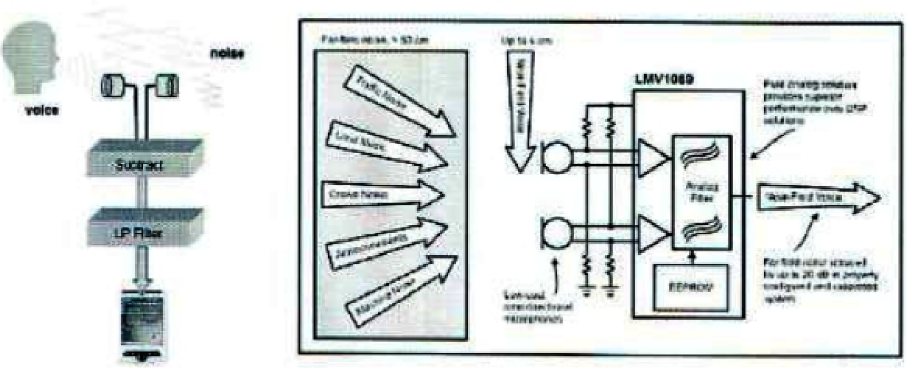


(a) (b)

Figure 39. Multichannel precision noise dosimeter

- *Speech recognition robustness improvement systems*

Another recent project involved the realization of a system for conditioning the signal coming from a microphone, to be used for reducing background noise in automatic speech recognition systems (ASR). These ASR systems typically use cheap microphones, since they run on notebooks, mobile devices etc.; solutions based on special microphone arrays are quite common, so that integrated chips for this special purpose are available on the market. ISTI-CNR lab proposed a simple solution using two small condenser microphones coupled together in opposite directions (figure 41a). The signal coming from the mike pointed towards the mouth is then combined with the other one using an operational amplifier, reducing background noise in a band interval dependent on physical microphone dimensions.



(a) (b)

Figure 40. Example of analog conditioning in a ASR system.

Figure 40b shows the structure of a chip (LMV1089) made by National Semiconductor, for environmental noise reduction. It is based on a reduced array configuration, based on only two microphones. With the microphones placed in the suggested configuration, this chip makes it possible to reduce the background noise up to 20 dB. The chip's internal conditioning solution is completely analog.

9. Conclusions

We have focused on basic principles, describing the main components that form a fully analog acquisition chain and that are also used in the digital chain. For the latter we described the operation of various types of AD and DA converters and the special requirements for analog conditioning needed to successfully perform the acquisition of audio signals. Finally, we showed examples of the use of certain circuit solutions for special requirements of audio recording, and recent projects in the Signal and Image Lab, also in specialized applications in the audio field. Along with the general principles, we have attempted to highlight the motivations of technological change aimed at overcoming the limitations of previous solutions, in order to achieve optimum performance levels in terms of requirements, consumption and small-size devices. As mentioned in the Introduction, the description does not pretend to be exhaustive, but can serve as an outline for further study. Following this criterion, we enclose a short bibliography.

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