IPAL

Integrated Powered Adaptive Loudspeaker

PCT Patent N°: WO-2008-018099 - PCT Patent N°: PCT/IT98/00031 - U.S. patent N°: US 6,281,767 B1

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What is IPAL?

IPAL ("Integrated Powered Adaptive Loudspeaker") is a patented technology for subwoofer applications representing the most effective tool available for the acoustic designer to achieve unprecedented results in terms of control over the sonic quality, SPL capabilities and overall system efficiency.

IPAL implements an innovative approach allowing to increase consistently the "mains input to acoustic output" efficiency path thru specifically matched amplifier, differential pressure sensing device, DSP and transducer combination.

BEHIND THE TECHNOLOGY

IPAL is the natural consequence of Powersoft's continuous effort for providing the Pro-Audio industry with high-efficiency solutions: despite the big step forward represented by Class D technology, it is clear that in order to overcome speaker's technology physical limitations and achieve real improvements in the acoustic performance, the overall system efficiency, rather than just the amplifier's, should be taken care of.

This is only possible through a much stronger integration between the amplifier and the transducer itself, and ideally realize an "active loudspeaker" with DSP capabilities allowing for a complete control over the acoustic output.

In the IPAL technology's development, the three factors above (integration, efficiency and control) were maximized by taking care of all aspects and components contributing to the system's performance:

- a novel approach to closed feedback loop has been implemented, based on differential pressure control method, allowing for extreme linearity and real time correction of the uncertainties of the acoustic system, as well as predictability and reduction of sensitivity from disturbing effects.

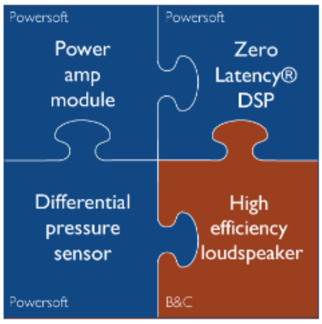
- specific design and energy optimized matching of the amplifier output stage and the transducer motor delivers an unprecedented value in terms of Acoustic Output to Mains Input Power Ratio.

- optimization of the transducer has been run to create a device that works efficiently at Large Signals and not at "datasheet" level, where the power demand sets constrains to the system.

These are some of the keys allowing an IPAL system to reach acoustic capabilities far above the expectations, and SPL and distortion performances unexperienced before.

What is an IPAL system made up of?

The hardware of an IPAL system includes four elements:



- a power amplifier module
- a differential pressure sensing device
- a "zero latency" DSP
- a specifically designed high efficiency transducer

The "IpalMod", provided by Powersoft, is the platform including the amplification section, DSP and pressure sensing system.

The power module is a single-channel Class D amplifier, with PFC-equipped power supply, able to deliver up to 8500W with IPAL speakers.
The module includes the Zero Latency DSP simm board.

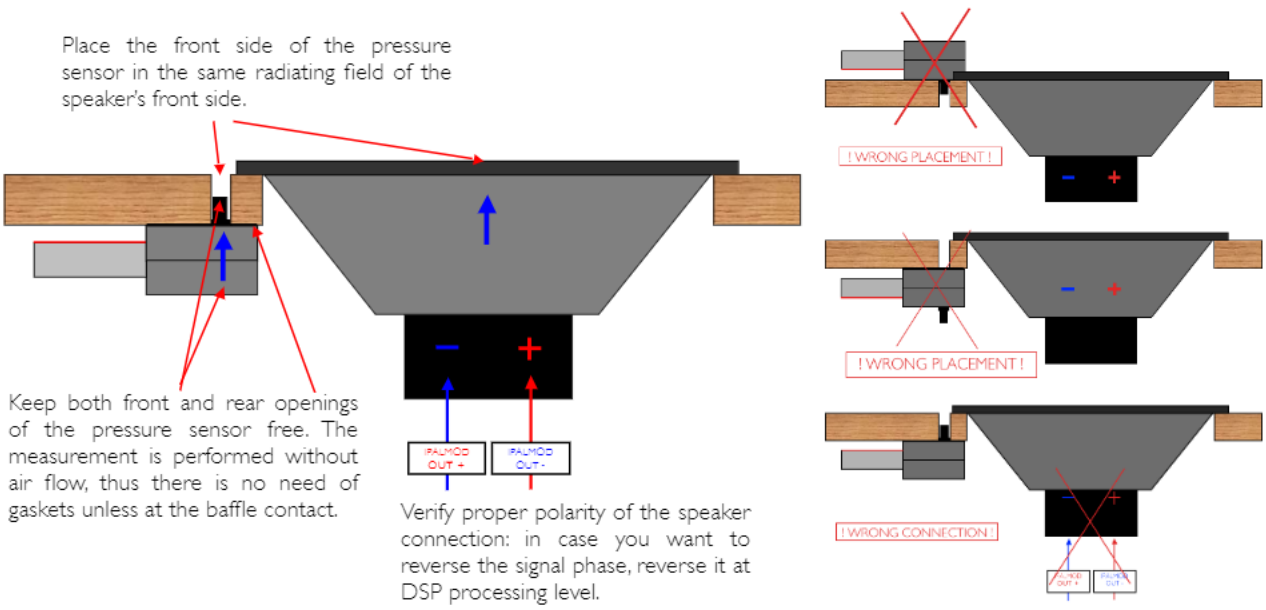
The differential pressure sensing device is connected to the electronics through a cm long flat cable for easy and proper placement on the cabinet's front panel.

An optional PCB board is available for input connection and serial ports for remote control / monitoring purposes.

B&C developed two different models (21" and 18") of IPAL compatible speakers, the 21IPAL and 18IPAL .The 21IPAL is intended to be used for single speaker applications, while 18IPAL is recommended for 2x18" configurations.

PLACEMENT OF THE PRESSURE SENSOR

The differential pressure sensing device should be placed as close as possible to the speaker's border, in a position where no turbulences are present inside the cabinet.



The IpalMod platform - Technologies

DPC – DIFFERENTIAL PRESSURE CONTROL

DPC® (Differential Pressure Control), is an innovative technology that is the core of IPAL control method. Full characterization of transducer and acoustical load conditions allows to correct in real time the uncertainties of any acoustical system with state of the art resulting performances. System linearity is guaranteed by a feedback correction that takes care of removing the limitations of physical transducers.

Building a global feedback method (that works) from electrical domain to acoustic domain has been an aim since the early days of acoustic, DPC® solves the problem in a elegant and, at the same time, extremely effective way.

"ZERO LATENCY" DSP

In any control method, a processing unit has to carry out calculations for correcting, filtering and generally speaking "processing" the signals received from the various sources. In the IPAL technology, this process is demanded to a DSP that has not only to be very powerful but as well "very fast": since it is impossible to correct what is already passed away, a specifically designed DSP core has been developed to minimize the In-Out latency of the system.

An innovative architecture that ensures an astonishing 10us (microseconds) latency on the critical feedback paths allows "analog type" feedback approach with the flexibility of a DSP core.

VIRTUAL SPEAKER MODELING

DPC® still allows an amazingly powerful tool to built a so called "Virtual transducer", giving the real transducer to behave accordingly to an "user defined" transducer , synthesized by the speaker designer, with a dashboard to manage the Thiele – Small or electromechanical parameters of the desired driver.

No alterations are produced by aging, power compression, production uncertainties on a device that relies its behavior on a mathematical model.

POWER FACTOR CORRECTED POWER SUPPLY

In the vision of a worldwide touring application, IPAL technology integrates in the power supply a PFC (Power Factor Corrected) High Efficiency Solution allowing a worldwide mains usage.

Benefits in current consumption and stability against mains variations are a plus in combination to an high voltage mains mismatching withstanding.

A Green approach dominates the whole design to contribute reducing the consumption as much as possible.

The IpalMod platform - Specifications

Power Module Specifications

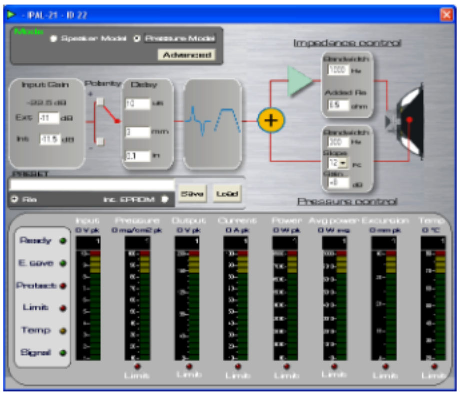
Power supply section		
	Topology	Switchmode Fixed Frequency PFC power supply
	Mains input voltage range	90 V AC - 290 V AC (allowed operational: 50 V AC/DC to 320 V AC/DC, 400 VAC tolerant)
	Power factor	> 0.95
	Average power consumption	400 VA
	Average efficiency	95%
Output stage section		
	High efficiency "Energy Cycling"	Switchmode Fixed Frequency Energy Recycling
	Output voltage (peak-to-peak)	390 V
	Output current (peak-to-peak)	240 A
	Peak output power	8500 W
	Average efficiency	95%

Processing and Control Specifications

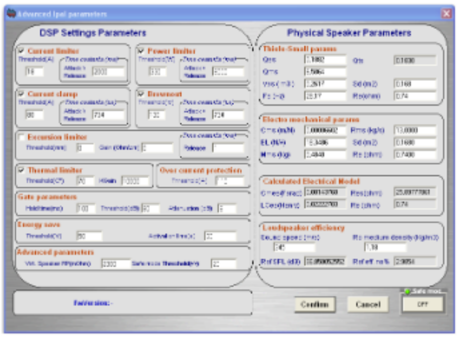
	Filter processing	1 low pass + 1 x high pass crossovers (up to 48 dB) Butterworth, Bessel, Linkwitz-Riley, 16 x PEQ
	Limiters and protections	Excursion limiter, current & power limiter, current clamp, clip limiter, Brownout limiter, thermal
	Meters	Input & output voltage, pressure, peak & average current, peak & average power, excursion, temperature
Virtual Speaker® Mode		
	Thiele-Small paramters	Qes - Qms - Vas - Sd - Fs - Re
	Electromechanical model parameters	Cms - Mms - Rms - Bl - Sd - Re
Differential Pressure Control® Mode		
	Impedance control parameters	Bandwidth, added Re
	Pressure control parameters	Bandwidth, slope, gain

The IPAL Software

The software developed by Powersoft for the IPAL allows to tune, process and monitor the system through four simple windows.

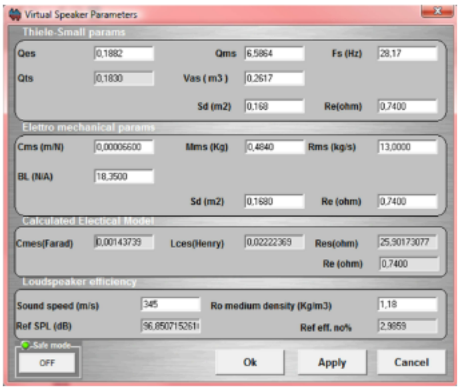


The main window allows to choose the operating modes (Virtual Speaker / Pressure Control), to set input parameters (Input Gain, Polarity, Delay), to manage the presets and fully monitor the behavior of the system thanks to 6 LED and 8 meters.



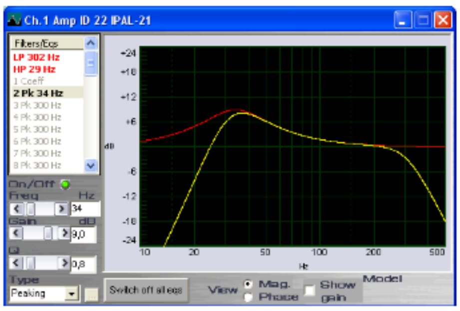
In the Advanced Ipal Parameters window the electro-mechanical parameters of the actual speaker driven have to be entered to allow the software to calculate the mathematical model.

IPAL is equipped with seven different protections (Power limiter, Current limiter, Current clamp limiter, Clip limiter, Thermal limiter, Displacement limiter, Over current protection): from this window all settings can be accessed.



In the Virtual Speaker Model operating mode, the physical speaker is driven to behave as one with different electro-mechanical parameters.

From the Virtual Speaker Parameters window's interface it's possible to set all of the parameters.



Of course, the IPAL software also have standard signal processing capabilities, through 18 filters: 1x Low Pass + 1 x High Pass Butterworth/Bessel/Linkwitz-Riley Crossovers (up to 48 dB) and16 x PEQ (Peaking, LoPass EQ, HiPass EQ, BandPass, BandStop, AllPass).

The B&C IPAL speakers - Technology

The 18IPAL and 21IPAL are two very high efficiency subwoofer drivers, based on two powerful neodymium magnets, designed specifically for use with the IPAL power amplifier technology and developed in cooperation with Powersoft.

18IPAL & 21IPAL MAIN FEATURES:

- Transduction efficiency
  The electromagnetic assembly of the IPAL speakers is based on two high energy, high density neodymium-iron-boron magnets, yielding a very high β factor $=(Bxl)^2/R_E$ (430 kg/s, for 18IPAL and 520 kg/s for 21IPAL) that provides extreme levels of electromechanical transduction efficiency and a very tight control over the moving assembly, as required by IPAL technology.

- Power handling
  The 18IPAL features a 116 mm voice coil with a 44 mm winding depth, yielding 1700W AES power handling; 21IPAL is capable of 2500W AES thanks to a 153 mm voice coil and a 48 mm high winding. The large heat exchange area keeps operating temperatures low, even in the most demanding applications.

- Mechanical robustness
  Both 18IPAL and 21IPAL cones are reinforced with carbon fibre and fully waterproof, in order to withstand the harshest environmental conditions.

- Excursion capabilities
  21IPAL reaches a full ± 22 mm (± 20 mm for 18IPAL) linear excursion capability (X_{MAX}).
  With a 1680 cm effective radiating area (1210 for 18IPAL), it can displace more than 7 litres of air in a single stroke (nearly 5 litres in 18IPAL).

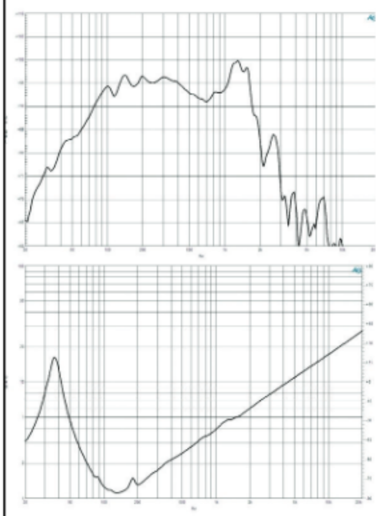
- Stability
  The voice coil of IPAL speakers are designed to ensure mechanical stability of the moving assembly in all operating conditions, and to minimize DC offset.

- Low impedance
  Low electrical resistance R_E (1,4Ω for 18IPAL and 0,7Ω for 21IPAL) keeps to a minimum the energy lost to heat by Joule effect and helps control the thermal dynamics of the whole system.

	18IPAL	21IPAL
β factor $=(B \cdot l)^2/R_E$	430 kg/s	520 kg/s
Voice Coil	116 mm	153 mm
Winding depth	44 mm	48 mm
AES Power handling	1700 W	2500 W
Xmax	± 20 mm	± 22 mm
Effective radiating area	1210 cm^2	1680 cm^2
Air displacement	5 litres	7 litres

The B&C IPAL speakers - Specifications

B&C SPEAKERS ®



21IPAL

LF Drivers - 21 Inches

5000 W continuous program power capacity
153 mm (6 in) split winding aluminium voice coil
37 - 1000 Hz response
99 dB sensitivity
Neodymium magnet allows a very high force factor and linear excursion
Double silicone spider with optimized compliance
80 mm peak-to-peak excursion before damage
Ventilated voice coil gap for reduced power compression
AVAILABLE TO OEM MANUFACTURERS ONLY.

Specification

Nominal Diameter	530 mm (21 in)
Nominal Impedance	1 ohm
Minimum Impedance	1.1 ohm
Continuous Power Handling	5000 W
Nominal Power Handling (AES)	2500 W
Sensitivity (1W/1m)	99 dB
Frequency Range	37 - 1000 Hz
Voice Coil Diameter	153 mm (6 in)
Winding Material	Aluminium
Former Material	Glass Fibre
Magnet Material	Neodymium Inside Slug
Winding Depth	48 mm (1.9 in)
Magnetic Gap Depth	18 mm (0.7 in)
Flux Density	1.35 T

Design

Surround Shape	Triple Roll
Cone Shape	Radial
Magnet Type	Neodymium Inside Slug

Design

Spider	Double Silicone
Pole Design	T-Pole
Woofer Cone Treatment	TWP Waterproof Both Sides
Recommended Enclosure volume	190 dm3 (6.71 ft3)
Recommended tuning frequency	33 Hz

Parameters

Fs	37 Hz
Re	0.7 ohm
Qes	0.22
Qms	4.9
Qts	0.21
Vas	155 dm3 (5.47 ft3)
Sd	1680 cm2 (260.4 in2)
EtaZero	3.2 %
Xmax	+- 22 mm
Xvar	+- 15 mm
Mms	487 g

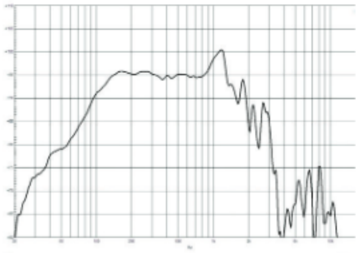
Parameters

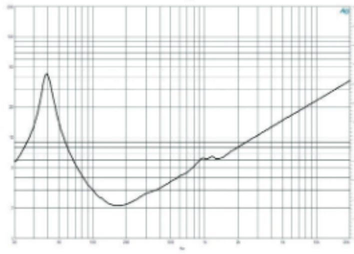
Bl	19.1 Txm
Le	0.5 mH
EBP	168 Hz

Mounting & Shipping

Overall Diameter	547 mm (21.5 in)
Bolt Circle Diameter	527 mm (20.7 in)
Baffle Cutout Diameter	508 mm (20 in)
Depth	270 mm (10.6 in)
Flange and Gasket Thickness	16 mm (0.63 in)
Air volume occupied by driver	16 dm3 (0.56 ft3)
Net Weight	22 kg (48.5 lb)
Shipping Weight	24.5 kg (54 lb)
Shipping Box	590x590x300 mm (23.2x23.2x11.8 in)

Service kit	TBA





18IPAL

LF Drivers - 18 Inches

3400 W continuous program power capacity
116 mm (4.5 in) split winding aluminium voice coil
32 - 1000 Hz response
97 dB sensitivity
Double silicone spider with optimized compliance
80 mm peak-to-peak excursion before damage
Neodymium magnet allows a very high force factor and linear excursion
Ventilated voice coil gap for reduced power compression
AVAILABLE TO OEM MANUFACTURERS ONLY.

Specification

Nominal Diameter	460 mm (18 in)
Nominal Impedance	2 ohm
Minimum Impedance	2.1 ohm
Continuous Power Handling	3400 W
Nominal Power Handling (AES)	1700 W
Sensitivity (1W/1m)	97 dB
Frequency Range	32 - 1000 Hz
Voice Coil Diameter	116 mm (4.5 in)
Winding Material	Aluminium
Former Material	Glass Fibre
Magnet Material	Neodymium Inside Slug
Winding Depth	44 mm (1.7 in)
Magnetic Gap Depth	12 mm (0.47 in)
Flux Density	1.5 T

Design

Surround Shape	Triple Roll
Cone Shape	Radial
Magnet Type	Neodymium Inside Slug

Design

Spider	Double Silicone
Pole Design	T-Pole
Woofer Cone Treatment	TWP Waterproof Both Sides
Recommended Enclosure volume	150 dm3 (5.3 ft3)
Recommended tuning frequency	35 Hz

Parameters

Fs	32 Hz
Re	1.4 ohm
Qes	0.14
Qms	4.2
Qts	0.14
Vas	164 dm3 (5.8 ft3)
Sd	1210 cm2 (187.6 in2)
EtaZero	3.3 %
Xmax	+- 20 mm
Xvar	+- 15 mm
Mms	330 g

Parameters

Bl	24.5 Txm
Le	0.65 mH
EBP	228 Hz

Mounting & Shipping

Overall Diameter	460 mm (18 in)
Bolt Circle Diameter	443 mm (17.44 in)
Baffle Cutout Diameter	422 mm (16.6 in)
Depth	242 mm (9.5 in)
Flange and Gasket Thickness	16 mm (0.62 in)
Air volume occupied by driver	10.5 dm3 (0.37 ft3)
Net Weight	11.9 kg (26.2 lb)
Shipping Weight	13.9 kg (30.6 lb)
Shipping Box	500x500x250 mm (19.7x19.7x9.8 in)

Service kit	TBA

Audio Engineering Society

Convention Paper

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Practical applications of a Closed Feedback Loop Transducer system equipped with Differential Pressure Control

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ABSTRACT

A closed feedback loop transducer system dedicated to very low frequency reproduction can be used in several different applications. The use of a feedback control loop can be very helpful to overcome some of the well known transducer limitations and to improve some of the acoustical performances of most of subwoofer system.

The feedback control of this system is based on a differential pressure control sensor. The entire system control is performed by a "Zero Latency DSP" application, specifically designed for this purpose in order to be able to process the system with real time performances. Practical applications to real world examples are being shown with design details and some test results.

1. INTRODUCTION

Low frequency reproduction systems that use Electro-dynamic transducers are based upon very well established approach from a long time. Substantial improvements in design, materials, manufacturing process have been achieved, but, despite large increase in performances, some weak points, due to the nature of the principle of operation still remain unaddressed.

Aging of the compliance of the suspension system, nonlinearities, trade-off between efficiency, performances and applicability in acoustic designs are a limiting factor for several low frequency loudspeaker designs. In addition to this, limitations in repeatability in the manufacturing process is another weak point, hard to overcome even in the most accurate production process. A very desirable low frequency transducer designed for outstanding performances could usually require electromechanical parameters that sometime could fight against the real feasibility. Moreover, this transducer could have, sometime, a limited usability in commercial designs since it could be not easy driven by commonly available amplifiers.

Modern technologies, both in amplifiers and in transducers engineering could expand the possibilities of low frequency system design. Generally speaking, the acoustic loading for a low frequency transducer is calculated and optimized upon the transducer Thiele and Small parameters in order to achieve a desired SPL acoustic response together with the achievable linearity performances and the intended bandwidth, and these performances should be easily obtained when connected to any generic power amplifier that acts like a voltage source.

Anyway, besides the ease of use of loudspeaker transducers when applied in the general approach there are also several limitations to accept that cannot be overcome.

We can resume some of these limitations as follows:

- Consistency between different samples
- Consistency during the lifetime
- Reliability when driven close to maximum limit level
- Suitability for any desired acoustic design
- Transduction efficiency.

A very good solution to overcome these limitations and improve the overall performances of low frequency loudspeaker systems could be the use of closed feedback loop methods to control the loudspeaker. This allows to correct and remove as much as possible the unpredictable and undesired effects of "real world " products, and paying attention to maximize performances and in increasing the system efficiency could also compensate for the cost of added complexity in the system design.

In the wide field of "controlled" transducer there can be found a lot of 'prior art'. Several different realization have been seen since the early 70's [1][2][3][4][5][6][7][8][9][10][11].

Among the various methods to realize feedback loop control, some of them are based on control of cone acceleration. This method has been able to obtain good performances in terms of linearity in closed cabinet designs, but for high SPL applications it does not show very good performances. Several commercial designs rely on such approach however, delivering consistent performances in terms of frequency response, transient response and overall linearity.

An alternative approach to realize feedback controlled loudspeaker could be based on the control of the cone speed. Methods based on speed control are also able to track a reference pre-processed signal that links the speed to desired output transfer function, still not allowing a complete freedom of applicability for acoustic loads different from closed box. Several other methods move around similar approach with limitations of applicability for acoustic loads that differ from closed cabinet.

We can say that one of the best methods presented in the past that features real substantial improvements in linearity is the *ACE Bass* system. This approach includes, in the closed loop, part of the electromechanical parameters of the physical transducer, being able to alter electrically some of those parameters thus being able to adapt the speaker behavior, within some limitations, to the specific acoustic design.

The ACE system is able to do that with some limitations:

- Inability to reduce the virtual moving mass of the synthesized system to values lower than the real one
- Inability to increase the compliance of the synthesized system to values higher than the real one.
- Being based on a "primary side" current feedback signal, the method can be highly affected by the nonlinearities of the physical transducer itself.

Therefore, the system above, even though it allows for several improvements in sound quality, does not leave a full degree of freedom in terms of synthesis options and, at the same time, does not fully account to the acoustic boundaries conditions that involves the transducer operation.

The novel approach proposed here bases its improved degree of performance on the realization of the feedback loop with Differential Pressure Control method.

2. THE NOVEL APPROACH

The presented system consist in a combination of an high-power, high performances Switching Mode Amplifier in conjunction with an embedded DSP that performs the double operation of both managing the loudspeaker system processing and taking care of the Differential Pressure Feedback Loop Control implemented on it. The described system can be matched to any high performances low frequency, high excursion transducer, even though its performances can be maximized if used in conjunction with some specific transducer that have been especially designed for it and that will be shown ahead in the paper. The system is referred as *IPAL (Integrated Powered Adaptive Loudspeaker)* [12].

One of the most important innovations of this novel approach is the possibility to use it as a configurable system that may be setup and applied to a wide variety of applications. The system can fit several different kinds of subwoofer loudspeaker and generic low frequency reproduction units of any kind. The system is packed-up in a sort of turnkey solution that once installed in a box, can be connected to a computer via a dedicated communication network and then configured, tuned and optimized for the specific application. Anyway, the configuration process can access to the manipulating of several different systems' parameters, but the approach to the system setup may be resumed in two different philosophies that are represented by two different operating modes.

2.1. Virtual Loudspeaker Emulation

The principle of operation of Virtual Loudspeaker Emulation is based on a double loop control structure. The inner loop takes care to built a very predictable and stable electromechanical structure that is able to make it work like a simple Voltage Input / Speed Output model, where the only visible transfer function from outside is the $B*l$ (Force Factor) of the physical motor assembly.

This very stable building block is then able to track a reference model that could be set at the user's wish and that would be the desired Virtual Transducer.

The Virtual Transducer parameters will be based on a numerical model and it is, of course, not prone to any modification over the lifetime.

The behavior of such defined structure is unfortunately unable to react properly to the acoustic boundary conditions, and the Input Voltage - Output Velocity relationship that would take place would be the one of the loudspeaker operating in the "empty space". In other words, the speaker behavior would be completely unaffected by the forces that an acoustic load acts on the radiating surfaces of the transducer.

The Differential Pressure Measurement, senses the difference of pressure between the front and the rear side of the radiating diaphragm due to the acoustic load, and uses this information to alter the behavior of the reference model, therefore according to the real boundary conditions.

This method allows the modeled unit to behave as a real transducer that keeps the desired properties of the numerical design. An additional description could be explained in the following figures.

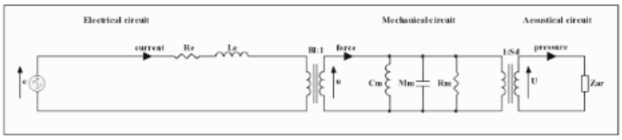

Figure 1 - Simplified Model of low frequency Electro-dynamic Transducer

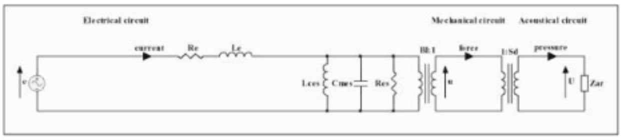

Figure 2 - Simplified model of a low frequency transducer with the Compliance, Mass and Losses characteristics moved to the primary side of the "*Bl to 1*" transformer.

As it can be seen in figure 1, a typical simplified model of a low frequency electro-dynamic transducer can be easily described in a schematic where all the physical domains can be involved: the electrical domain together with the mechanical and the acoustical domain. The most meaningful values taken into account are Voltage and Current in the electrical domain, Force and Speed in the mechanical domain, Pressure and Volume Velocity in the Acoustical Domain. As shown in figure 2, is possible to bring from the secondary side of the *Bl:1* transformer to the primary the most important mechanical parameters: Compliance, Mass and Losses transposing them into their electrical equivalent parameters *Lces, Cmes, Res* modeled respectively by an Inductance, a Capacitance and a Resistance. If a series of negative resistance and a negative inductance respectively equal to *–Re* and *-Le* (figure 3) will be added at the generator output, it is possible to remove from the primary side of the *Bl:1* transformer all the elements of the parallel resonant circuit formed by *Lces, Cmes* and *Res* and the synthesis of the circuit could be represented in figure 4 where parallel resonant circuit has been eliminated and the active elements with negative values added in series can be easily realized with the contribution of a negative impedance amplifier whose output impedance behavior can be modeled according to the desired *–Re* and *–Le* values.

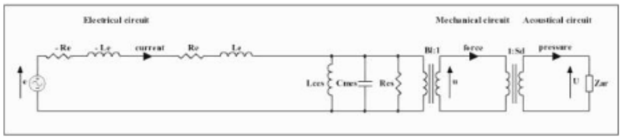

Figure 3 – Simplified model where a series of negative Resistance (*-Re*) and a negative Inductance (-Le) has been installed into the primary side.

The whole primary side of the *Bl:1* transformer has been reduced to a very desirable block where the input signal is feed to a simple buffer amplifier that drives the *Bl:1* transformer with a theoretical unlimited current capability. This arrangement basically allows for the diaphragm speed and its relative Volume Velocity to be very predictable and fully independent from the forces that act on the surface of the diaphragm itself, and link the diaphragm movement directly to the input signal.

The demand for pressure, force, and current, imposed by the acoustical boundary conditions in which the transducer diaphragm operates are therefore accomplished by current capabilities of the Voltage Buffer Negative Impedance Amplifier at the primary side of the *Bl:1* transformer.

Very simply, a voltage input signal corresponds to a speed of diaphragm at the output, and the relative proportionality is given by the *1/Bl* , were *Bl* is the force factor of the real transducer.

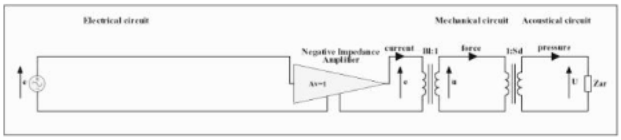

Figure 4 - Simplified model where at the primary has been eliminated the resonant parallel circuit and a Negative Impedance amplifier realize the negative *–Re* and *–Le* output impedance characteristic.

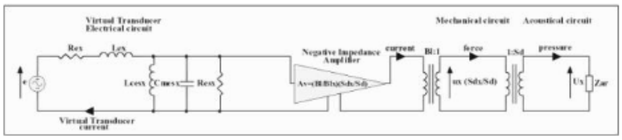

Figure 5 – Simplified model with a set of desired virtual parameters included before the Negative Impedance Voltage Buffer.

As shown in the above figure 5, it's possible now to define a set of "virtual" electrical parameters of a desired transducer to be emulated, here defined as *Rex, Lex, Cmesx, Lcesx, Resx, Blx, and Sdx*.

Giving the Voltage Buffer Negative Output Impedance amplifier a constant gain structure as *(Bl/Blx)(Sdx/Sd)*, it's easy to demonstrate that this equivalent structure will behave acoustically as a transducer with electrical parameters equal to *Rex, Lex, Cmesx, Lcesx, Resx, Blx* and *Sdx*, where *Sdx* and *Blx* are the Diaphragm Area and the Force Factor of the Virtual Transducer.

Therefore, as already mentioned, the only limitation of this synthesized loudspeaker model is that it does not react to the boundary condition (acoustical pressure) that is acting on the diaphragm surfaces of the real transducer, and in fact, such a model arrangement will behave identically, in terms of Volume Velocity/Input Voltage relationship, in any acoustic loading conditions: empty space, free space, open air or cabinet acoustical load.

Sensitivity of the newly arranged system will be similar to the sensitivity of the desired Virtual Transducer that is being synthesized, despite the actual motor strength and the actual radiating surface featured by the real transducer.

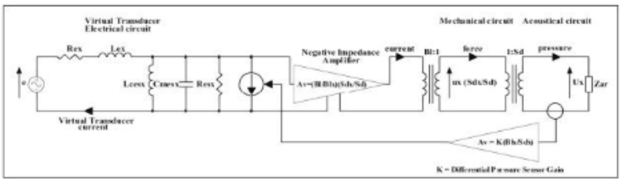

Figure 6 – The complete system model including the feedback loop with the Differential Pressure signal

It's necessary, at this point, to rebuild a relationship between the forces that actually act on the diaphragm surfaces and the input reference model.

A direct measurement of the difference of acoustical pressure acting on the front and rear diaphragm surfaces, weighted by K, the differential pressure sensor gain and by the ratio of the virtual force factor and virtual diaphragm surface is fed back to the emulated transducer model.

A controlled current source proportional to the above value is here able to rebuild the backward effect of the boundary conditions on the reference model.

It must be clear, once again, that:

- *Re, Le, Cmes, Lces, Res, Bl* and *Sd* parameters are referring to the real transducer,
- *Rex, Lex, Cmesx, Lcesx, Resx, Blx* and *Sdx* are parameters referring the Virtual transducer to be emulated.

u and *U* are Speed and Volume Velocity referring the real transducer and *ux* and *Ux* are Speed and Volume Velocity referring the Virtual transducer to be emulated.

Moreover *Zar* represents the acoustic radiation impedance and *K* is the differential pressure sensor gain.

2.2. Global Pressure Control

The novel feedback loop controlled loudspeaker system also allows for an alternative control method. This method involves user's substantially different approach if compared with the already described Virtual Transducer Modeling.
This alternative approach still relies on the differential Pressure Control and permits a very large degree of freedom in using the system from user point of view.

Given a system structure as described in figure 7, it's possible to use it to make it operate in order to match a specific target pressure response and use the output differential pressure measured at the surfaces of the moving diaphragm as a control signal. This value resembles the overall acoustical response with exception of the pressure/SPL transfer function that correlates these entities. Being the SPL output of any acoustic signal derived from the relationship between pressure and acoustical impedance, it's straightforward to use a direct measurement of such a pressure signal to make the system track a reference target pressure model.

A closed loop structure that comprises amplifier device, transducer and acoustical influence is therefore realized with a good stability. Benefits in terms of predictability of the system response and reduction of sensitivity from disturbing effects could be substantially appreciated.
This approach allows for a consistent method to make the physical system to perform accordingly a defined Target Pressure Model Response set by the user.

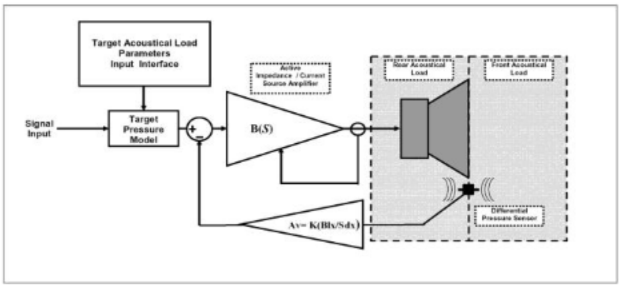

Figure 7 - Block Diagram representing the novel system configured for Target Pressure Model tracking

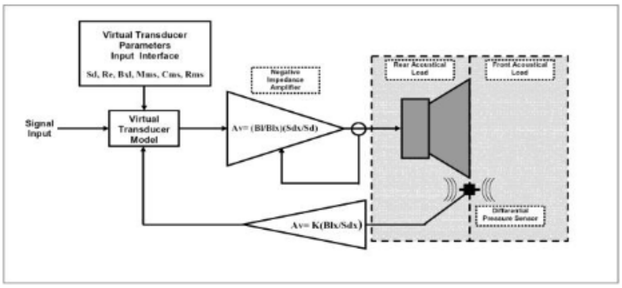

Figure 8 - Block diagram of the complete system while used in Virtual Loudspeaker Mode

3. SYSTEM APPLICATION

One of the most important innovations of this novel approach is the possibility to use it as a configurable system that may be setup and applied to a wide variety of applications. The system can fit several different kinds of subwoofer loudspeaker and generic low frequency reproduction units of any kind. The system is packed-up in a sort of turnkey solution that once installed in a box, can be connected to a computer via a dedicated communication network and then configured, tuned and optimized for the specific application. Anyway, the user during the configuration process can access to manipulate several different systems' parameters. The approach to the system setup may be resumed in two different philosophies that are represented by two different operating modes.

3.1. Virtual Speaker Model operation Mode

As described in the previous section, one of the possible ways of using the described system is the so called Virtual Speaker Parameter mode. This way of using this feedback controlled Loudspeaker/Amplifier system is the practical implementation of a way to operate that allows the user to configure the *IPAL* system transducer for a specific desired set of parameters. As already mentioned, previous attempts to built practical realizations of feedback loop controlled loudspeakers were also designed to do that. The novel approach here basically consists in having represented the inner speaker model with a numerical model inside the DSP. A block diagram of the complete system is represented in figure 8.

The inner DSP model of the speaker can be, in fact, numerically modified in order to make the speaker appear to the user to behave at the user wish. Moreover, the differential pressure feedback control allows for the modification of a wide set of parameters, and since the speaker model is a numerical model, this suggest to realize the user interaction via a computer interface that allows him to modify the loudspeaker model and adjust the speaker parameters. This also permits to use the transducer parameters like additional "Tuning Parameters" while optimizing the system setup. Setting the transducer parameters in order to get the desired acoustic response from the loudspeaker system is a completely different action from applying some specific equalization, because the parameters are implemented involving the differential pressure feedback loop and then subjected to the loop tracking.

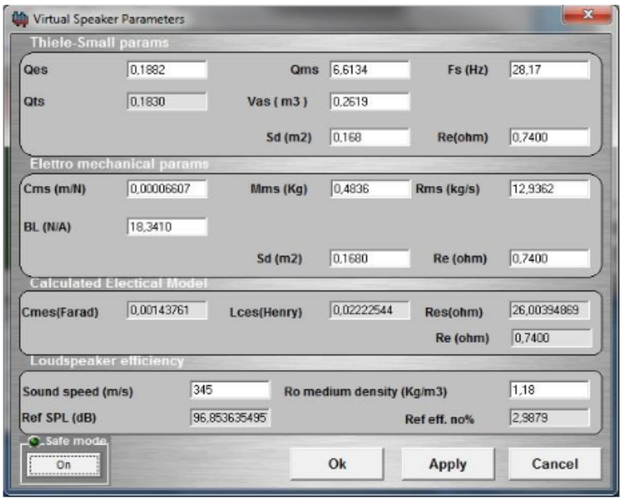

Figure 9 - Software interface where the user can set the desired target parameters

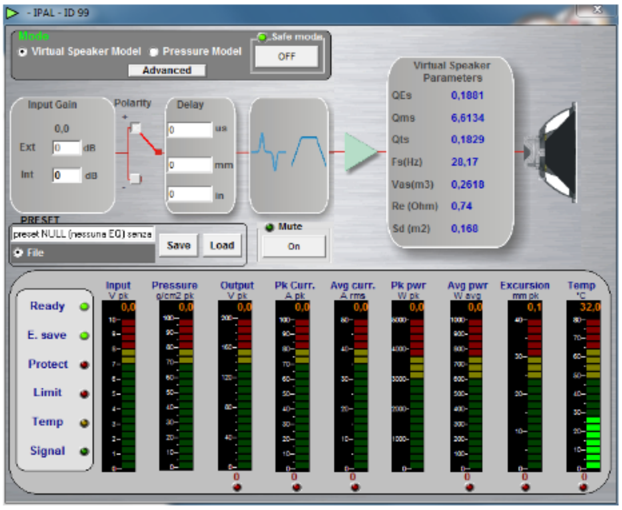

Figure 10 – Software control panel interface while operating in Virtual Speaker Model mode

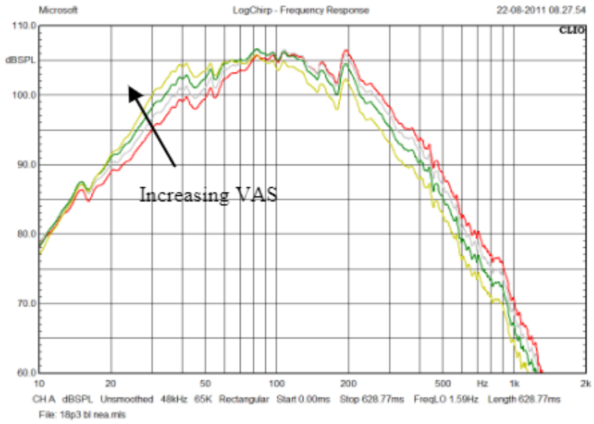

Figure 11 - Effect of increasing *VAS* of the Virtual Loudspeaker Model from the real value of 261 Lt. to 500, 800, 1000 Lt.

Some simple examples are shown in figures 11 and 12 to visualize the effect of modifying the values of two simple parameters on the Virtual Speaker Model. Figure 11 shows the effect of modifying the Loudspeker *VAS* from the value of 261 Lt. (Actual *VAS* of the physical speaker) to 500, 800 and 1000 Lt. Figure 12 shows an example of the effect of modification of the Virtual Speaker Bl value from 18.3 (actual physical value) to 15, 12 and 9 Tm.

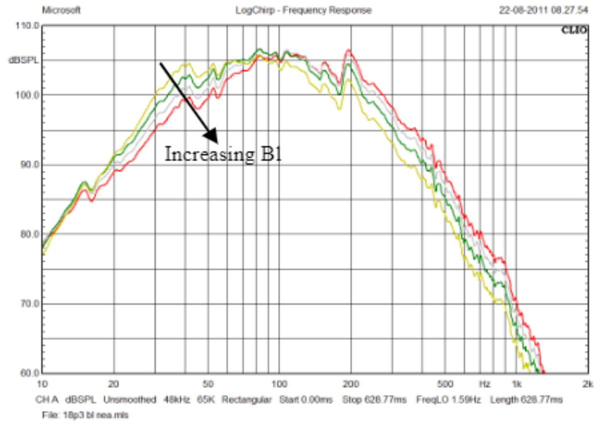

Figure 12 - Effect of decreasing *Bl* of the Virtual Loudspeaker Model from 18.3 (real value) to 15, 12, 9.

3.2. Pressure Model operation Mode

As already explained, pressure model operation is an operation mode in which the *IPAL* system uses the Differential Pressure Signal and a system control variable. The system control panel changes its appearance as can be seen in figure 13 and a specific interface that is shown in figure 14 allows the user to set directly the influence of the control variable on the system operation and set, ultimately, the amount of feedback control that should be used in the system.

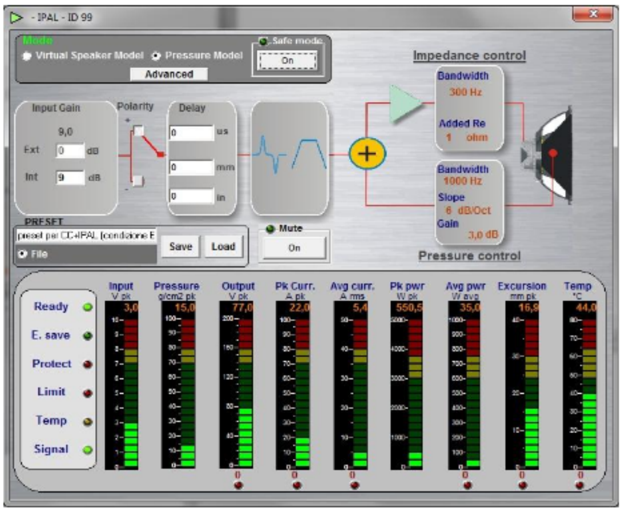

Figure 13 - Software control panel interface while operating in Pressure Model mode

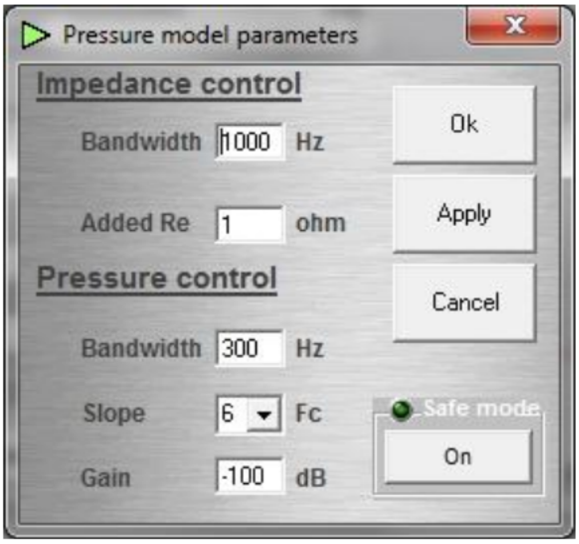

Figure 14 - Software interface section where differential pressure control loop parameters can be accessed

As it can be seen from the above picture of the dedicated control panel, the user is able to set the amount of Feedback Loop Signal from a minimum level of -100dB (feedback off) to 0dB and over, also setting the bandwidth limit above which the control signal should start to be reduced from the nominal value. The amount of feedback loop that may be acceptable depends from the specific application and the specific design where the *IPAL* system is being set. The upper limit of the amount of feedback will be determined by the system Phase Margin Limits and it must be carefully set in order to let the system operate in stable and reliable mode under any condition. From the Pressure Model control panel it is also possible to set the amplifier output impedance. This feature also permits the system to adapt the loudspeaker transducer to any specific acoustical design. The output impedance could have a bandwidth of operation as well. The optimal setting of these parameters must be still carefully set under the respect of phase margin requirements in order to get the necessary stability of operation.

Once the system has reached the desired level of control with the appropriate and safe stability conditions, the final target response that the low frequency system should have could be reached with appropriate equalization and filtering that may be applied in the input equalizer via a dedicated user interface as showed in figure 15.

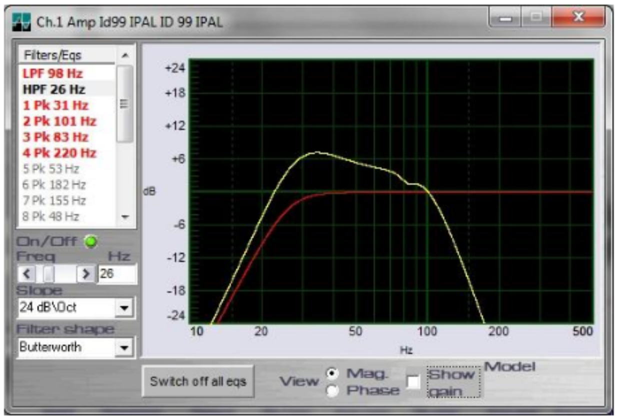

Figure 15 - Input DSP software Eq Panel interface

In the following pictures there are reported some interesting test results that have been gathered from a 18" transducer optimized for *IPAL* mounted in a common direct radiating bass reflex loudspeaker box having a volume of 120 Lt and tuned at 36 Hz. There are some 4 cycles tone burst measurements that have been placed at some relevant frequency points: 25Hz, on the first resonant peak on the loudspeaker system impedance, 36 Hz where was the tuning frequency, 50Hz where approximately the second resonant peak was located and then 80Hz and 120Hz, two frequencies of the upper bass range, one located well inside the system bandwidth while the second represents the upper usable frequency limit.

The following diagrams report the test results captured with digital oscilloscope reporting at the same time the input signal on one track and the output pressure measured in front of the cone surface in the other. Not all of them are using the same vertical scale, and this can easily understood and taken into consideration.

From the following measurements it can be observed the effect of the use of the Differential Pressure Feedback control loop, used in Pressure Model mode. The benefit in terms of control of the cone movement is evident and appreciable to the upper frequency limit.

It must be noted that the behavior of this subwoofer without the control loop activated represents a general behavior of such kind of loudspeaker systems, very common to the most of bass reflex subwoofers. Time response behavior of many subwoofer systems is, in fact, quite similar to the system used in the example.

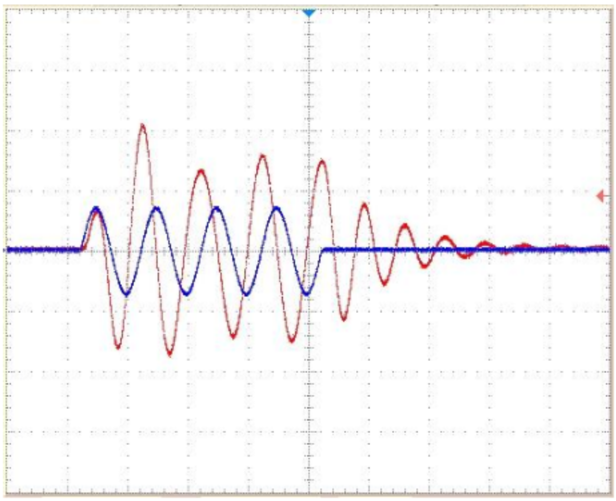

Figure 16 - 25Hz, 4 cycles, sine burst input signal with NO Pressure Control Loop

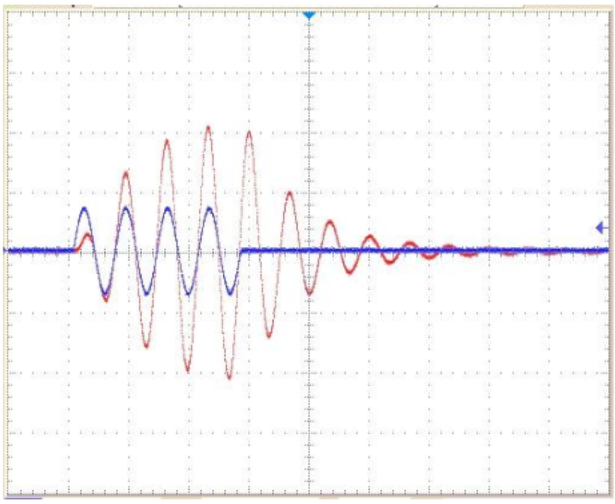

Figure 18 - 36Hz, 4 cycles, sine burst input signal with NO Pressure Control Loop

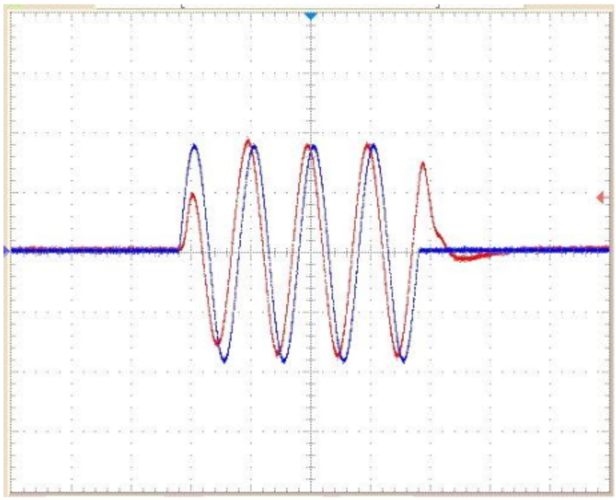

Figure 17 - 25Hz, 4 cycles, sine burst input signal with Pressure Control Loop

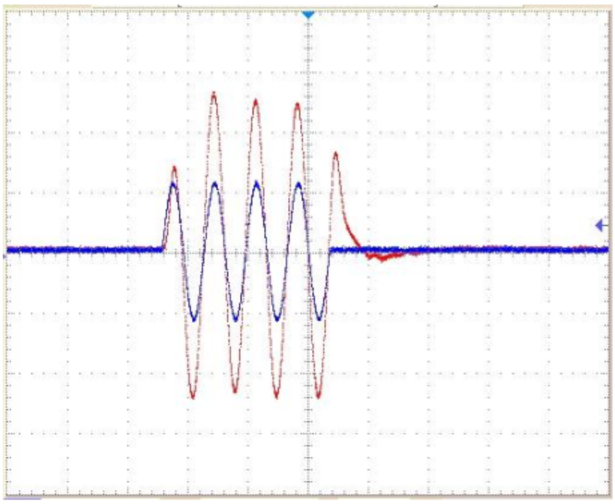

Figure 19 - 36Hz, 4 cycles, sine burst input signal with Pressure Control Loop

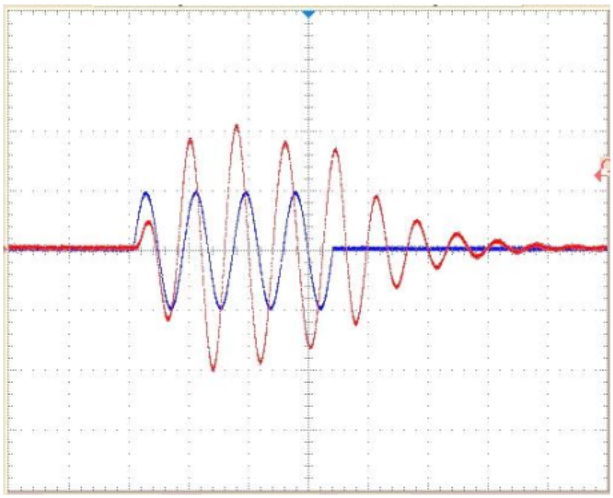

Figure 20 - 50Hz, 4 cycles, sine burst input signal with NO Pressure Control Loop

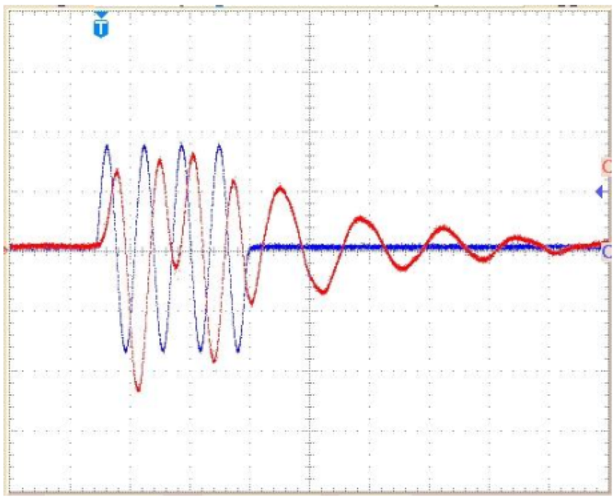

Figure 22 - 80Hz, 4 cycles, sine burst input signal with NO Pressure Control Loop

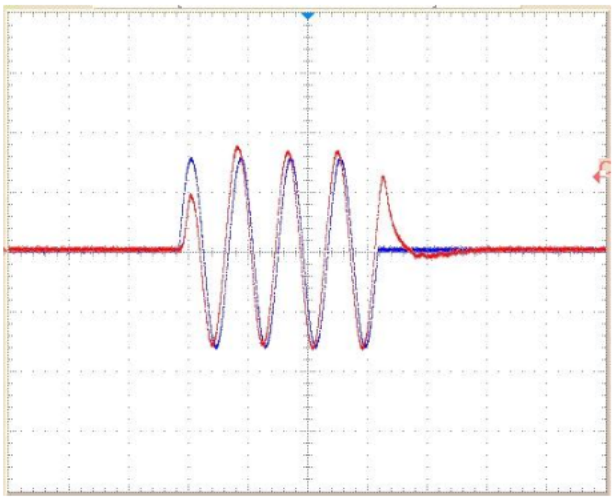

Figure 21 - 50Hz, 4 cycles, sine burst input signal with Pressure Control Loop

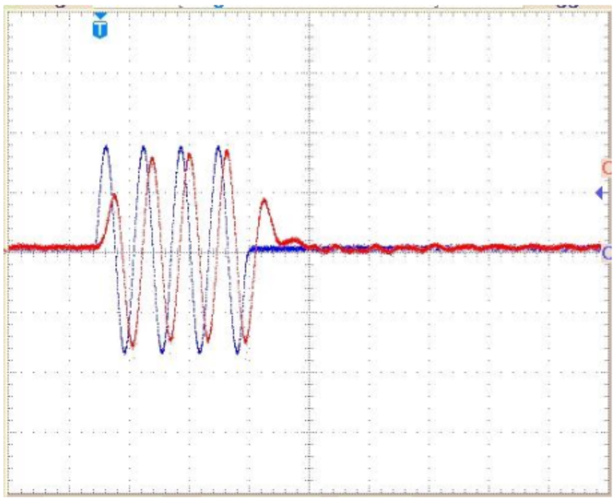

Figure 23 - 80Hz, 4 cycles, sine burst input signal with Pressure Control Loop

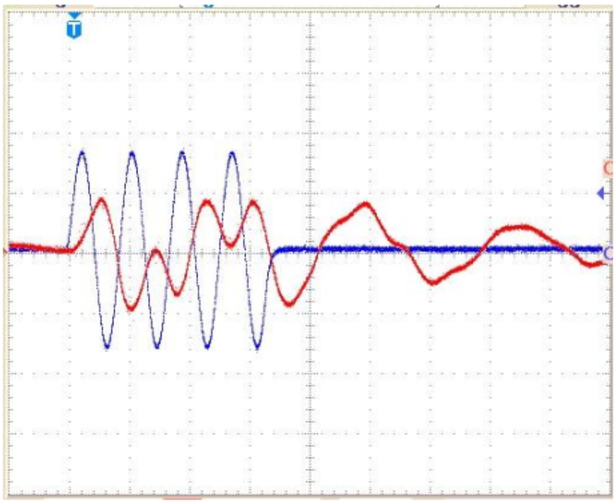

Figure 24 - 120Hz, 4 cycles, sine burst input signal with NO Pressure Control Loop

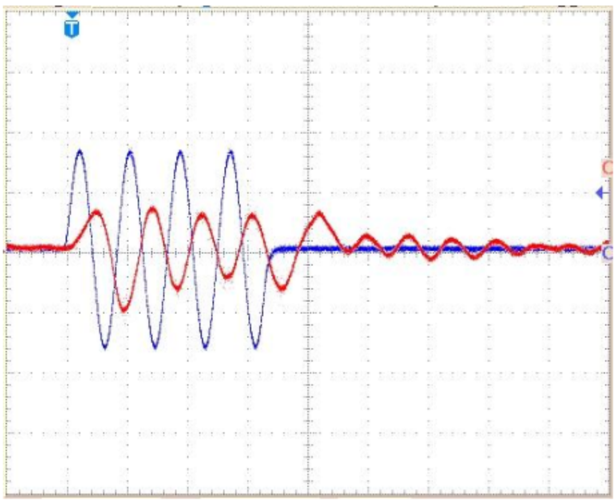

Figure 25 - 120Hz, 4 cycles, sine burst input signal with Pressure Control Loop

3.3. Example of transducers designed for the Differential Pressure Feedback controlled system

It could be of some interest, for those interested, to mention some details about sample loudspeaker transducers that have been designed and optimized to maximize the system performances. Particularly, these sample loudspeaker transducers are manufactured by B&C Speakers.

Even though this system can be matched to many different kind of loudspeaker, it is very important to point out, anyway, some of the relevant features that a transducer should have in order to maximize the benefit of being coupled to this novel feedback controlled systems.

At first it could to be noted, in fact, that one of the most important advantages of using a switching amplifier stage, usually referred as "Class D" amplifier, is the ability of this kind of power amplifier to manage any kind of Voltage to Current relation at its output, up to the full quadrature between them, practically managing a fully reactive load without any problem. Moreover, where the load is very reactive, the power that is bounced back from the loudspeaker to the amplifier output, in Class D operation, it could be easily recovered and recycled into the power supply rails. This kind of situation would be not possible to be managed by a traditional analog output stage instead, because the power dissipation would be extremely high in that case.

The logic consequence for this is that the loudspeaker to match to the system must be a speaker whose characteristics could be maximized in terms of overall efficiency making it features a very high "Motor Strength" and then an inherently high efficiency loudspeaker that will exhibit a very much reactive electrical behavior. If this last characteristic would be maximized, the amplifier would mostly manage reactive power that is bounced back and forward between the amplifier and the loudspeaker not creating substantial real power dissipation, increasing the overall power efficiency.

Moreover, a very high motor strength over a log excursion capability it would be essential to keep efficient control on the voice coil and cone movement. Some very good transducer to mach this feedback loop controlled loudspeaker would be then shown here. They represent some fine example of use of this system.

The following loudspeaker parameter sets represents very fine example of 18" and 21" loudspeaker that are specifically optimized for this system and allows for maximizing the overall performances.

Figure 26 - IPAL 18 Sample Loudspeaker

18" Sample Loudspeaker Parameters:

Fs	47 Hz
Re	1.37 Ohm
Sd	1210 cm2
Vas	84 Lt
Qms	6.3
Qes	0.23
Bl	22.15 Tm
Mms	283g

Figure 27 - IPAL 21 Sample Loudspeaker

21" Sample Loudspeaker Parameters:

Fs	28.2 Hz
Re	0.74 Ohm
Sd	1680 cm2
Vas	261 Lt
Qms	6.59
Qes	0.19
Bl	18.34 Tm
Mms	484g

3.4. System design considerations

Being the system heavily relying on the inner loop effectiveness, a very predictable and "flat" *Bl* vs. displacement characteristic of the used transducer is very desirable. Thus a specifically designed motor structure should be a major requirement, even though not mandatory.

Moreover, taking into account that a proper active cancellation is necessary, and being the *Re* and *Le* in the inner loop assumed as constants, is therefore necessary that these two parameters would be very stable in the used transducer.

Although some correction of *Re* vs. temperature is viable, these parameters should be to be varying as less as possible in order to avoid the necessity of direct temperature measurement of the voice coil or the tracking of it with a sophisticated thermal model.

Both the above constrains are achievable in a transducer that has been optimized for very high efficiency, having a $(Bl)^2/Re$ value that is above the usual ranges.

Differential Pressure sensing is also a critical point, since the effect of the acoustical load is derived by a "single point" measurement. At low frequencies, were wavelength of reproduced signals are sufficiently larger than the physical dimensions of the transducer itself, and most of the dimensions of the acoustic system in which the transducer is installed it is relatively easy to keep the distance from the location of the sensor to the centre of emission of the primary transducer relatively small (respect to the shortest operating frequency wavelength). In this case, negligible errors are introduced by the pressure control loop. However, upper frequencies where of the physical transducer behavior is influenced by directivity or by other high order phenomena, are not to be considered as range of operability of the system, or, at last, it could not be required that the system behavior will strictly follow the synthesized model.

It's to be considered anyway that the whole system relies on a "lumped parameters circuit", and consequently its behavior is subjected to some obvious limitations of such assumption. It has to be said that under normal operating conditions like the installation in a relatively compact subwoofer system or into a low frequency unit, the model assumptions are usually widely satisfied.

Figure 27 - *IPAL* Amplifier Module with DSP inside

Figure 28 – Differential Pressure Sensor

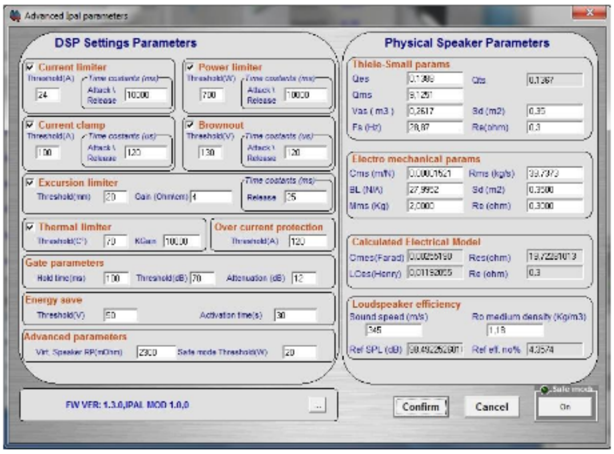

Figure 28 –A view of the inner system setting parameters interface where also physical transducer parameters are set

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