When and How does the grouped R1 work ?

why is so important?

When the Roundffusor1 are used as the sole room treatment [any kind of room], even if we remember or not the Shannon theory of information [where 0 means non or lost information and 1 being the transmitted and understood /perceived message - for us the meaning of a song's words and context and the music's details] all room acoustical imperfections are or tends dynamically to me corrected. With the R1, what more than 99% of listeners perceived is an obvious clearer sound without any trace of any kind of echoes (echo cancellation obtained in a purely acoustical way), the overall room time coherence containing all the details from the source.

The 25-35 msec interval where the named preceding effect arrives and makes distinct two or more reflections is "embedded" and masked into the diffusion field. The loudness [one of it's definition being "the perceived sound intensity when we are focusing our attention towards the sound"] is freely and suddenly perceived even by the hearing impaired people. The estimated 1,5-2 units of loudness increase is a clear explanation of R1's adaptability and a confirmation of the diffusion superiority against the old and too tired absorption. More, a direct result of using the R1 is a perceptible EDT's reduction, becoming sensible smaller then the RT60. It is known that the EDT is considered as the subjective way for us to perceive the character of a room along with it's reverberation (RT) timbre and "RT's tails" (which for the EDT - Early Delay Time, is a kind of RT10). If EDT is bigger or similar with the true RT60, then the room's intelligibility is reduced and along with it, and the musical details, covered by the reverberation time's envelope.

Shorter EDTs generally mean better communication and clearer & natural recordings, since room falls off more quickly and masks the information to a lesser degree. Those facts may be expressed also in this way: "The increase in the scattering coefficient decreases the amplitude of early reflections, time-smears the non-specular scattered energy, and changes the relative balance between direct, reflected, and reverberant energy" (from Edge Diffraction and Surface Scattering in Concert Halls: Physical and Perceptual Aspects, Rendell R. Torres, Nicolas de Rycker and Mendel Kleiner). It's exactly what we indicated (more than one year) earlier, on the left diagram, our conclusions being similar with the above ones with the addition that from a clean, increased diffusion, the direct field's content is more dense, real life's timbre - so no traces of coloration.

Along with the goal of a shorter EDT, we must be able to measure some parameters related with intelligibility and process the results into a score (see Fig.5) that correlates with how a human listener might judge the intelligibility. This parameters are: a) the Speech Transmission Index or STI (the results are processed into an intelligibility score that ranges from zero - bad to one - excellent and b) the

Percentage Articulation Loss of Consonants (%Alcons)- positive numbers are desirable. In all rooms treated with Roundffusor1, the EDT was smaller than the true RT60.

At the recent International Conference of Romanian Society of Acoustics on Sound and Vibration 2004, held in Bucharest / Romania , after the presentation of the three newly corrected Radio station's studios of the State Romanian Broadcasting Co. using our Roundffusor1, from the immediate round table discussions, resulted once again the difference between the actual state of the art of a practically constructed recording studios -with the specific subjective conclusions and the obvious inertia of strictly applying the actual acoustical standards and interpret ting the results analogous with the standards. And the subjective appreciations (in our 3 cases) were enough different from what "should" be the "letter" of some standards. In the work presented, were shown the acoustical measurements before and after the R1's mounting upon the acoustical materials found there. In the R1's situation, some standards have to be revised. Was a clear recognition, that our R1 is a huge step forward in acoustics.

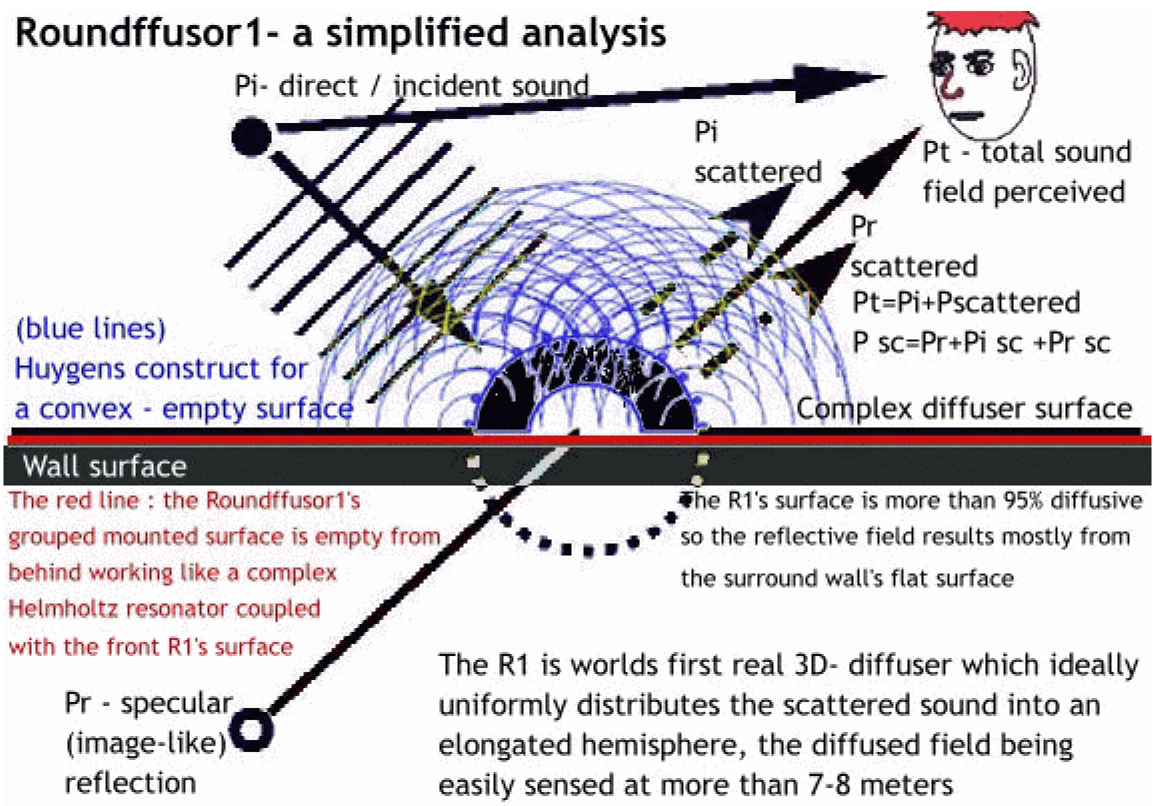
Bellow is a R1's simplified analysis (adapted from Torres et al 2004). Intuitively, the Huygens constructs (initially introduced to explain light scattering) from the whole convex / semispherical complex surface will results in an elongated- in phase with the source- orientated hemispherical 3D highly uniform scattering wave lobe. Please note the coupling between each R1's back cavity with the one below and above [the coupling between the single Helmholtz resonator -each R1], all this in parallel columns or rows [depending on the supports direction toward the ground]. Thus, the horizontal diffused lobe field is at a perpendicular angle with the resonating /filtered field emerging from the grouped R1's complex cavity.

Sort of R1's reduced thickness, the diffraction on the lateral wood's supports side is very low but on the other two open sides , a kind of additional scattering might appear. This scattering of diffused field sound by the filtered sound from the complex R1's inside one, might produce transitional /constructive interferences related to the frontal incident wave field which is the music- a dynamic field. The measured phase of those two perpendicular fields are similar and without irregularities. This, correct in a way the known bibliography speaking about phase curve deformations.

As seen at "R1's critical measurements" (at R1'1 Tests) each vertical of mounted R1's column acts differently towards the local eigenfrequencies. Internal R1's filtering means that the R1's cavity self resonant frequencies interact with the harmonics of a sound wave passing though each columns. Basically, the result is that the harmonics from the incident field near the resonant frequencies are enhanced, while those away from them are damped [remember the organ situation]. This is for a normal situation. In the R1's case, it's adaptable filtering means that this complex internal interaction map the musical field, correcting it in a musical way. As the musical instruments stays such a long time in their traditional shapes, the same for the R1 which may be considered a musical instrument...

■The complex Helmholtz resonator which is formed by the back geometry of the grouped R1 are enough similar with the air resonances of closed-box musical instruments were all kind of resonances play a much more active role in shaping the sound. Air modes – resulted from resonance related to the direct sound in the R1's case - can have a pronounced affect upon how acoustic energy flows through the entire system, causing subtle changes in the volume and sustain of the vibrations induced in the soundboard by the strings, and ultimately, the timbre radiated to our ears.

■The resonating air behind the grouped R1s as the air inside a real musical instrument has many modes of resonance, all occurring simultaneously, all filtering and shaping the final sound we hear. The more the space is divided up into smaller sub-spaces, the more resonant frequencies there are, all interacting with one another to create a complicated and unique mixture - a musical one. That's why the R1 are so musical and best suited for any kind of music .



More, from the paper ''Influence of surface scattering characteristics on the sound quality of reverberation'' of Jacob Mueller, Mendel Kleiner, Ning Xiang, and Rendell Torres there are some subtle results for the late reverberation, even if not as clear, indicating that the "individual" sound of scattering surfaces will influence primarily the sound quality of the early part of the reverberation.

when does the grouped R1 work ? why is so important?

This must be just another reason on why our R1 is so efficient and subjectively performant sort of its unique shape. Seems that our brain along with the auditory system searches the most sensitive part – the early delay time (EDT) of the reverberation time and is searching for pleasant or not minute tails. Inside of the named EDT, the qualitative EDT's content influences our senses in the first time/ moments (in fact in the first 100 msec) when we hear music, sound generally. To be more precise for a listening room with a typical RT60 = 0.4- 0.5 seconds, the brain is immediately aware about the sound quality from the first or dynamic time parts of maximum 100 ms (or 0,1 second) each. All this dynamic time parts are panned in 3D and each room at a certain moment as a sole overall EDT, a kind of room's fingerprint. A recording, even filled with plenty of electronic reverberation contains this ''fingerprint'' or a statistical time hologram which may be recognized between two notes or words. Seems that the R1 ''manages'' to make musically the link between the smaller dynamic time parts and larger sound events scales, musically meaning the feel of natural sound, the natural timbre.

I believe that the grouped R1's surface acts like a double acoustical antenna, working in small/ imperceptible fractions of time, in and out, like the ''Stargate's'' opening in the well known and made TV serial. The R1's back interior column's air vibration [designed to calm the room's resonances], ''dictates'' in phase to the front diffusion surface / room's diffused field. This complex antenna have a high gain adaptable (scattering intensity) to the source's level/s - difficult but not impossible to demonstrate/prove (the distributed scatterer such as turbulence or,... a Gaussian distribution of the amplitude of initial field) being -perhaps- the manifested loudness increase in the room sensed even at 12 meters away from the R1's grouped surface. That's why we choose a short word ''best'' to analyze some rooms. This presumption must explain why even room off-center pair of sound sources are perceived on room's axis with an astonishing wide image sort of the R1's active high gain, the result - to be found everywhere in the room, being a statistical time/frequency like hologram, this ''particle'' information containing the stereo mix in all details.

We are aware that other researchers believes that combining periodical with a-periodical geometrical shapes on their mounted acoustical materials, the non-uniformity diffused lobes encountered in all theirs materials having periodical shapes will be somehow minimized.

What we know for sure is, that in the Roundffusor1's case - which is a highly symmetrical so of a periodical when grouped mounted shape, the resulted 3D diffused lobe is at the limit of the polar plots ideal / perfection. Analyzing in detail the Schroeder's RT curve's slopes, they were plotted almost 99% parallel in all octaves, meaning that the sound field was indeed being produced diffuse (highly or perfectly, as you prefer) by the grouped R1s so the measured RT 60 or any kind of RT, was conform with the theory. In addition, if we mount an R1, on both lateral walls near the listening position by changing their heights equally, one may vary the depth and the height of the perceived sound, meaning controlling at will the sound stage. This confirms, once again, that the Roundffusor1 is in a totally new category, a giant step in the art of acoustical materials.

There are also some opinions from academics, such as Dr. Roger Penrose of Oxford University, who argue that brains do not work in a way comparable with a computer, so any kind of simulation of the brain that is built on digital architecture and uses traditional programming techniques is doomed to failure"...

{ such things are impossible ? }

when does the grouped R1 work ?

I believe that our brain considers those small parts of sound context (duration ≤ EDT and the JND - just noticeable differences - related with all sort of differences in aural perception resulted from some 17 cm distance between our ears) and dynamically is comparing them with our memory - educated musically or not.

Referring to the non digital way of working of our brain, there is no big surprise that many people really loving hearing music keep and enlarge their LP's collection and at the recording studio level, main path electronics are using tubes / valves (considered "analogues") or software emulating old /vintage valve gear- especially compressors.

The "brain's" road from 100 ms down to around 2 msec. I spoke about he qualitative EDT's content influences our senses in the first time/ moments (in fact in the first 100 msec) when we hear music - sound generally, then about the 25-35 msec (milliseconds) interval where the named preceding effect arrives and makes distinct two or more room reflections, then about the first direct sound path of 2,5 - 3 msec from one loudspeaker to the shortest path - supposed an observer, then about the continuum - tympanum / inner ear / nerves / brain. But, when does the grouped R1 work ?

Here is my idea , after so many years of searching it. Why I insist so much about symmetry in the room? Like that we have a field "ready" for the original field of the recording, there will be minimal interferences between them. With the help from the R1's diffused field, your CDs or LP or movies will sound like the original ones, with their perfection (meaning the reality at the recording's moments and space) or imperfections. A blend of music lovers dreams with the real high fidelity.

And all this, is done practically simultaneously with the incident sound, because our hearing system cannot perceive...when. Don't forget that the tympanum have equal air pressure above and below, the same with R1's surface. This "when", in my opinion is the human perception's delay. Is the time of Summum bonum of all firings in the continuum - tympanum inner ear nerves brain and all the (neurons or cells) firings takes some time something under 2 msec or less..

This "time" depends on room dimensions and source/s position/s. Might be considered a human constant but depends slightly on aural human ability obtained from education or selective listening from live or recorded music. From all musical parameters, the timbre is predominant and this cannot be perceived so fast because an active listening is needed and this requires an active memory use, a comparative one. It is true that all those "sensations" are synchronized in around 100 ms when "what we hear" is "compared" with what we know and we are "entering" in the "like / dislike" moments being "locked" in our brain along with the music.

Simply said- in the interval of around 2 msec to 1 sec we begin to "understand" what we hear. Under around 2 msec, we hear NOTHING.

There, inside this time of around 2 msec, the R1 "manage" to be active or beginning to be. We CANNOT sense nothing or almost nothing inside this sensorial delay. So the said...real time, is not quite real time, even for sound neither for vision... Just be happy - as me - that the R1 is...in time.

Even when we have less than half back wall (behind the loudspeakers) covered with R1, we have an evidence that using the R1 the acoustic field is "forced" to symmetry, or ... the "expected" wrong stereo image becomes a wide natural well defined stereo.

In all rooms treated with R1 (even if the walls are partially covered with initially mounted absorptive materials) are in conformity with the EBU tech3276 & 3286 recommendations at the EXCELLENT level and beyond it. In our opinion, what matters most is the resulted room's field timbre & sound dynamics, music's or voice's natural pattern.

■Just that being "deaf" for a moment (around 2 msec) is not enough...There are some other ideas (commented by me here) that spatial hearing is inherently non-linear implying that we might have to consider some nonlinear storage format, which only helps store transients more accurately (the brain's in/out connection with reality). From here thinking to a (perceptual) temporal resolution of an audio system approach tending to infinite (temporally - meaning memory not frequency domain which - resolution - is distorted from the very tympanum - "first sound touch".

■Also - a myriads of implication: inherent nonlinearity in hearing (computational models of microcilia!) used to explain difference tones, perception of harmonic/near-harmonic spectra, missing fundamentals - levels of pattern recognition (learned vs. intrinsic) - comodulation of masking release and profile perception as signs of cross frequency band processing at a low level - the consequent refutation of strictly tonotopic place theories of pitch. A ... conclusion?

■What if ALL this non linearity is just another sensorial delay? Lets consider our recording versus loudspeakers / room interface. When a pure diffused field is created in the room then inside the sensorial delay enters the diffusers actions. Consider and might be 99% true that this sensorial delay is the "real time" (or time "0" - zero) for our hearing. Mathematically and using the concept of quantum mechanics (because we have a human observer / consciousness where the loudness versus timbre or spectral envelope are subdued as 'beliefs', and other thoughts are again delayed - Stapp, 1972), the absorptive material is adding a zero + something (x) depending on its surface relative to the room dimensions. Saving the World from absorption must be everyone's "crusade".

■If we define the ideal impulse perception as the one at the moment zero, then , having absorption in the room, we get all the time a "zero + x" delay. We know that in music the impulse is the attack time of a note, like the one that we hear very very near a piano or better very near to a drum set, with just a hard played note. The more absorption we have the more far away from the piano / drum we feel. Can we consider R1 as a case solved within the "mind over mater and back to the mind" concept ? I think yes. There is no person experiencing the R1's sound and going back to a previous situation.

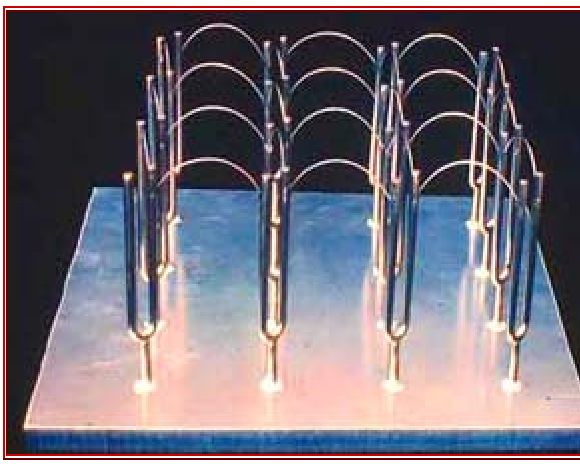
■Non linearity in sound perception might be considered a shifting in all parameters so keeping or succeeding a pristine room acoustic situation helps our brain in less "computing" and more musical pleasure. In an absorptive environment he have also pleasure (statistically, for the non connoisseurs) but is the pleasure of LOW - FI (read Low Fidelity). Be sure that a decent Hi-Fi system sounds more musical in a proper acoustic environment than a very expensive system in an absorptive one. Being a real music lover is the true answer. We can sense and pleasure the music? then we are gifted creatures..

Why I insist so much about symmetry in the room? Like that we have a field "ready" for the original field of the recording, there will be minimal interferences between them. This symmetry is not only a geometrical one. The R1's surface will produce a consonant environment for your recording and room. This consonance or unison of the musical fundamentals / octaves or their harmonics details (exact or even approximate) are continued in our brain as neural-firing coincidence. This is the ideal continuum interface - recording / room / hearing and is the way a musician perceive the music. With the help from the R1's diffused field, your CDs or LPs or movies will sound like the original ones, with their perfection (meaning the reality at the recording's moments and space) or imperfections. I prefer to expose here this rough series of ideas.

Similar results and ideas but obtained in the laboratory conditions are here. Headphones listening is not exactly similar to real room music listening, but doing such type of research in a room using two loudspeakers, is extremely difficult. Why? Even 1 degree of head movement, change our aural perception. Like that, the statistic repetitivity between the subjects is broken.

Intuitively, the grouped R1 can looks like that, a finite numbers of coupled musical tunning forks as local oscilators, as a demonstration of wave propagation in two-dimensional arrays of scaterers. The detail, the important detail, is in how many directions to be coupled the musical tunning forks and the fact that in the R1's situation, we have two layers of complex coupled forks tied together by the Janus effect.

Besides the WHEN history, we have also a HOW ones. In the project presented below, we have a 4000 Kg of 6 ways of totally horn driven audio system, where we designed two very low frequency of 12 meters long double horn drivers working from 1 to 100 Hz. The whole system, using 50 R1 on the back wall and 24 R1 upon the ceiling has a resolution of +/- 0.25 dB from 5 Hz to 20.000 Hz.




A blend of music lovers dreams with the real high fidelity

Important note from the PCT International search report for our patent

1. Classification of subject matter and fields searched : IPC 7 G10K11/20 E04B1/84

2. The documents considered to be relevant for this invention- the Roundffusor1, were all in the "A" category [document defining the general state of the art which is not considered to be of particular relevance, having almost nothing in common with a previous art]. There are 3 older patents [1939,1957,1983 , with indications, where appropriate, of the relevant passages, also, not considered to be of particular relevance] and a <u>sole paper to be open from here.</u> This indicate the almost absolute originality of our invention. The paper is an indication of our predictions that the Roundffusor1 may be used and as Music hall's ceiling stage reflector or generally in big Music Halls because is the most efficient diffuser on the market, it's diffused field being easily perceived at more than 12 meters from any angle.

The Roundffusor1 may be used in any kind of rooms

Recording studios

Post production studios

Broadcasting studios

Listening rooms

Home theaters

Quality control rooms

CD mastering

Press conference rooms

Film mix and dubbing stages

Music Schools & practice rooms

Churches