EARO

High Definition Audio

EAROLOGY

In the past few decades the majority of the HiFi industry has, with a few notable exceptions, focused loudspeaker design on achieving a technical performance which is not consistent with the psychoacoustics related to our hearing. One could say they have not been on the right track.

To successfully create speakers that offer a reasonable opportunity to accurately convert the electrical signal to sound requires understanding of human auditory perception, acoustics, mechanics and that of both analog and digital electronics.

Dimensions

One of these focal points of the industry has been to focus on achieving a flat frequency response. Whilst a reasonably level frequency response is important as it relates to the colouring (or tonality) of sound, it is of a lesser priority to something occurring in another dimension, namely *time domain*. A loudspeakers ability to follow the acceleration and deceleration of an electrical signal is fundamental to the subjective observations such a “realism” “clarity”, “detail” and “presence”. It also has significant impact on both stereo imaging and perception of depth. Time domain, or more correctly phrased, temporal response is thus the speakers ability to follow musical transients.

More scientifically expressed, a speaker is an information device. A steady tone carries no information as it has no bandwidth. Due to its predictability, each cycle is exactly equal to the previous one, it is thus of little use as method of quantifying loudspeakers performance. Yet, it is this the that appears to be regarded as the key parameter. A steady tone is a singular dimension and has no component in the time domain. A transient on the other hand is not predictable and thus has bandwidth as it occupies the time domain, bandwidth means that information is propagated.

As a small indication of how a large part of an industry has lost the holistic view during it´s focus on frequency response issues, a not insignificant parameter in the audio production chain may have been missed . As recently suggested by industry expertise (Watkinson) a transient is, contrary to a steady tone, an asymmetric signal. Thus it has a polarity. Yet, there is no mechanism in the reproduction chain to identify this and ensure its correct polarity at playback. In practice, there is a 50% chance that the speaker may move inward at the onset of a transient whilst in real life the actual transient caused a positive pressure front.

Our auditory system uses transients as part of identification of a sound. It is a primeval instrument for survivability. Is it far away or so close that it is a threat? In principle the human auditory system relies both on the time of arrival (including phase) and intensity to locate a source of sound. Our brains, *given undistorted cues*, has an astounding ability to process complex sound and pick out relevant parts. As an example we all know about the “cocktail party effect”. That is, how to pick out conversation in a crowded and loud environment relies on ability to match up the time difference of sound arrival to each ear.

Many musical instruments make use of transients, instruments are mostly played by humans expressing emotions and transients convey them. This is why Earo devotes special attention to its speaker designs ability to reproduce musical transients and to be phase accurate throughout the audio spectrum. To achieve this, we have applied various technologies to produce the desired results. Speakers with a *high definition* capability exist but are rare, not all are able to appreciate what such a device offer as *most listeners are attracted to a sound*, not a transparent medium. But given time and enough source material anyone, not just experienced professionals, begin to appreciate the purity in reproduction as it reveals the art behind the recorded music.

The listening room

Strictly speaking, it is unachievable to accurately reproduce music in a different room from where it was recorded in. This issue becomes even more convoluted as it may be argued that accurate reproduction is not possible at all as a speaker *propagates* out information picked up from a microphone that sensed sound *arriving* to it.

Luckily, it is possible, with skill and very good tools to create an illusion of a musical performance in the room it was recorded in. Therefore, we believe in not trying to achieve the unachievable but to focus on the parameters that make speakers accurately reproduce the signal conveyed to it. Be it good or poor. The rest, the musical experience follows from this. But beware, this coin also has a flip side to it, it will also reveal what you have not heard before.

The listening room has a huge impact on the reproduction. A speaker in one room will have noticeable different tonal quality in another room. The conclusion is one that the room plays a role. In fact it plays a very significant role. If you are serious about creating a realistic illusion of musical performance being made in your home, you have to not expect more than the imperfections of a room. Fortunately there are many tools available today to both identify and correct for imperfections in a room. However, you cannot correct for all inferiorities in speakers any more than you can correct a Donkey to become a Thoroughbred.

Earo produces speakers that may be placed into two categories, direct radiating and omni-directional. The latter makes use of room characteristics and produces a relatively dispersed and uniform soundfield. Whilst there is increased depth some of the clarity of definition is lost. The direct radiating models do the opposite, the soundfield is narrow but also highly controlled and accurate. A relatively narrow listening position is produced.

Distortion and listening fatigue

Tonal quality is a term used in assessing a musical instrument. Basically it is the complex structure of harmonics produced by the instrument under varying playing conditions. In a reproducer of such instruments one strives for the opposite, complete freedom of tonality. The speaker must not add or remove any harmonics, neither in level, phase (time) or temporal dimensions. This is very difficult to achieve but with good engineering practice it is possible to get reasonably close. In fact, satisfactorily close.

Many technologies have been explored in attempts to find the optimal transducer and whilst there are both good and bad results the perfect technology has not been invented yet. Some succeed in one dimension and fail in another. As in engineering in general, it is usually a matter of making a range of compromises. The challenge is to make the right ones and makes these from the start.

To name a few, such compromises are related to esthetical design, cost, availability of components and materials, production methods but also to skill and artistry. A wise individual ones said that simplicity is the ultimate in sophistication and this neatly defines one of Earos core values.

A basic requirement in order to create the illusion of realism is to be able to reproduce the same sound level as if the instruments where in the room and the listener at same distance. In many cases this produces a type of listening strain commonly referred to fatigue. Listening fatigue is a reaction to a sensory overload to sound that the brain attempts to, but cannot, interpret. The reaction to the strain is a desire for silence. Even if subjective and not readily quantifiable, it is a very good indicator of the overall performance of the HiFi chain.

Whilst the recent decades of technological achievements in principle has made the recording and storage medium totally transparent to the point that there is no scientific foundation to debate “quality” in this chain, there is one last mile that has not made quite the same progress. Namely, the speaker system.

Speakers are the last mile in the reproduction chain and it so happens that it is here that most of the distortion occurs. Earo has addressed this by applying the involved physics in its designs in such a way that distortion is significantly reduced. Oddly, the theory behind this has been known for nearly one hundred years. A possible reason why so few products reflecting this have not been produced may be found in the theoretical and physical complexity involved and the aforementioned compromises.

We have taken a long and hard look at this and applied the most recent technology to make the necessary compromises in the right place. This is reflected in our products appearance and functionality.

Loudness

Perception of the strength of a sound is termed loudness. The meaning of the word is deceptively simple but as we will see is not so. Most natural sounds from instruments have a natural level or at least range of sound levels.

Harvey Fletcher and WA Munson demonstrated already in 1933 that our perception of loudness varies with frequency and that this in turn varies with the sound pressure level. Thus, how we interpret a sound is a three dimensional set of parameters. Add to this that there is a variance on this from one individual to another and it may be understood that we are dealing with a complex matter.

This varying perception effect is so apparent that we are almost not reflecting upon it any longer, as we turn the volume down both the low and high end disappears. In reality it doesn´t, its just our perception of what we hear that changes.

Reducing this to the context at hand, sound reproduction, it follows that there is in fact a proper loudness at which a record shall be played in order to achieve greatest fidelity. In fact studying the data from the now standardized ISO 226:2003 one may note that the curve begins to linearize (flatten) at sound pressure levels of 80-95dB. This is a level where the performance played appear to be as strong sounding as if they where occurring in the room. Avoid listening at levels higher than 80-90dbSPL as you are entering the range of possible permanent damage.

So we now understand that playing too low will make the sound unbalanced, what about louder then?

Here comes another fundamental effect into action. Our ears produce brain stimuli by nerve ends attached to hairs on the basilar membrane. This membrane is a rolled up wedge shape “tongue” vibrating in a liquid. The membrane responds to sound by resonating in a particular place corresponding to the frequency. The issue here is that the higher the level the broader will the part of the membrane vibrating become. It follows from this that as the level increases the ears resolution in frequency falls. The term for this is called “critical bandwidth” and is the underlying effect exploited, not always well, in audio compression systems.

In practical terms, as you play louder than the ideal level, your ability to discern details in the music is reduced. That’s why poor musical live performances frequently are masked by high sound levels.

Understanding this fundamental relationship, it becomes clear that the speakers must be able to produce those levels with dynamic range to spare and with low distortion in order to be able to offer proper reproduction.

This also gives an idea of what kind of music is best used, or avoided, in judging quality of your sound system.

More about our ear

As mentioned above, the basilar membrane rests in a fluid. How are then the vibrations of a sound transferred to this liquid? This occurs in the middle part of the ear, behind the eardrum is a set of levers performing impedance transformation. What is particularly important is that there is a feedback (closed loop) gain control. This is implemented by fine muscles under brain control tensioning such that the inner ear receives the correct range of levels. This is the mechanism responsible for the great sensitivity range possible, allowing time for adaptation. We now have introduced another dimension; our perception of sound is also a matter of our state of mind.

Phase and time

As mentioned above, time (or phase) is a key parameter in the auditory system to identify sound. A musical instruments fundamental tone has a rich and complex set of harmonics. These have a very precise relationship in phase and time. Yet, of the entire chain, from musical performance to recording and out of your amplifiers speaker outputs, the time/phase relationship is commonly retained, only conventional speaker technology breaks this integrity. What is worse is that they do so by design.

Earo strictly believes in keeping this integrity to be a key parameter in loudspeaker design. For larger parts of the audio spectrum time/phase integrity is a not just an important parameter, it is a fundamental one. Only at the extremes of the audible range, where our ears precision begins to fade, can compromises safely be made. In fact we believe in a strong link with how phase affects subjective judgment in clarity and realism as well as positional accuracy in space.

This is why you will not find that our speakers consist of multiple elements, spread out two-dimensionally over the front of a rectangular speaker baffle. Multiple elements may at first appear to be a solution of the problem of tonal quality and broad sound field but the practical implementation creates more problems than it solves. One of them is the separation of harmonics in space and time causing inaccurate reproduction. In fact it trades off one problem to create another one (beaming versus polar response).

Another effect of locating elements on a flat surface of a box is the smearing of the stereo image due to what is known as edge diffraction. This shows up as reflections at every edge of that box, confusing the listener as to where a sound comes from. Whilst a box must have edges, the more elements on the baffle the more smeared will this soundfield become.

Earo is dedicated to resolve this conundrum and our designs reflect this. Most of our speakers use fullrange drive units that manages to resolve greater parts of the musical spectrum. Only where perfection in tonal neutrality is required, may more than one transducer be employed. In all such cases the physical implementation is such that no phase errors are introduced.

We call this philosophy for ***Earology.***

A note on recorded material

Live, live studio and studio recordings

For decades, when the recording technology with difficulty could be moved, the musicians came to the studio. Multitrack recorders allowed for practical work by separating individual musical performances in time and even worse, in space. In other words the musicians where not playing at the same time or in the same place. The rhythm track was laid down only with those musicians in the studio, later vocals where added and later other instruments. Two very important things were lost. Firstly the abstract effect of humans making music in unison was removed and replaced with a sanitized feel as the musicians performance was "corrected" by an engineer. Secondly the search of this controllability also removed the spatial dimension naturally present in live recording even if multi-microphone techniques where used. Such studio made products could sound good under very skilled recording and mixing hands but more often it was not. Why? Simply put, the process of separating each instrument also separated the interaction of room acoustics. We eventually learned that it is better sounding, or more correctly sounding, to have any room acoustics rather than not having one at all.

This has now evolved into studio techniques that tend to produce live or at least near live performances in a controlled acoustic environment. Such recordings have clearly gotten the spark of life back into them, not just because of the spatial reappearance but also because the musicians are really playing together.

Live performances on the other hand have improved much. The challenge is a real time combined PA and recording situation where the objectives do not match. Some live recordings have occasional technical flaws such as to tight miking resulting in momentary distortion, but the overall musical performance, even including the audience, usually excels making the flaws less significant.

When listening to Earo speakers you should find they bring out this realism in live and studio live performances whilst the limitations of old school recordings also become quite obvious.

Earo - designed from function