ACOUSTIC ORCHESTRA DEDICATED MACHINES

AODM
CLASS 0

5 REASONS WHY AN AODM SYSTEM WILL ALWAYS GIVE YOU GREATER SATISFACTION COMPARED TO ANY OTHER "HI-END" SYSTEM

AGAINST THE STANDARD AUDIO

SHORT HIGHLIGHTS

AODM

AODM is an acronym for Acoustic Orchestra Dedicated Machines and defines a series of technical standards that equipment must meet to be certified as AODM.

Equipment built according to the specific AODM standards optimizes performance for the reproduction of acoustic music, namely music made using acoustic musical instruments. This includes classical and chamber music, Jazz, Soul, Country, and particularly human voices, or singing.

From a technical standpoint, the AODM postulates represent technical solutions where linearity and transparency are essential sought-after qualities in every product that aims to be certified.

AODM Standards

- No capacitors in the signal path *
- Inductive power supply system
- No feedback between stages (This also applies to Integrated Operational Amplifiers, which are not allowed)
- Class A circuit architecture or Pure Symmetry
- No crossover within the frequency range of 100-6000 Hz (Exclusively referring to speaker systems)
- The presence of capacitors and/or transformers in the signal path determines the class to which the certified AODM device belongs

CAPACITORS IN THE SIGNAL PATH

Capacitors are 'indispensable' components in all electronic circuits, particularly in amplifier circuits for audio signals. They are used to separate the direct currents powering the circuits from the alternating currents generated by the audio signal itself.

Capacitors traversed by the audio signal introduce losses to it due to three essential factors:

- Capacitive reactance, which introduces increasing losses as the audio frequencies decrease. The greatest losses are particularly observed at low frequencies

- Internal resistance due to leakage currents, which introduces additional losses across the entire audio frequency range

- Memory effect – the worst of all – as it introduces losses on the weaker portions of the audio signal, effectively eliminating the finer details originally contained within it

For the reasons we've just seen, it is essential not only to minimize or even completely eliminate capacitors on the audio signal path, but it is of utmost importance, in order to best preserve musical quality, to use capacitors of the highest possible quality.

In terms of losses related to the so-called memory effect, the best capacitors are, in order:

- Oil and paper capacitors; the absolute best for audio applications

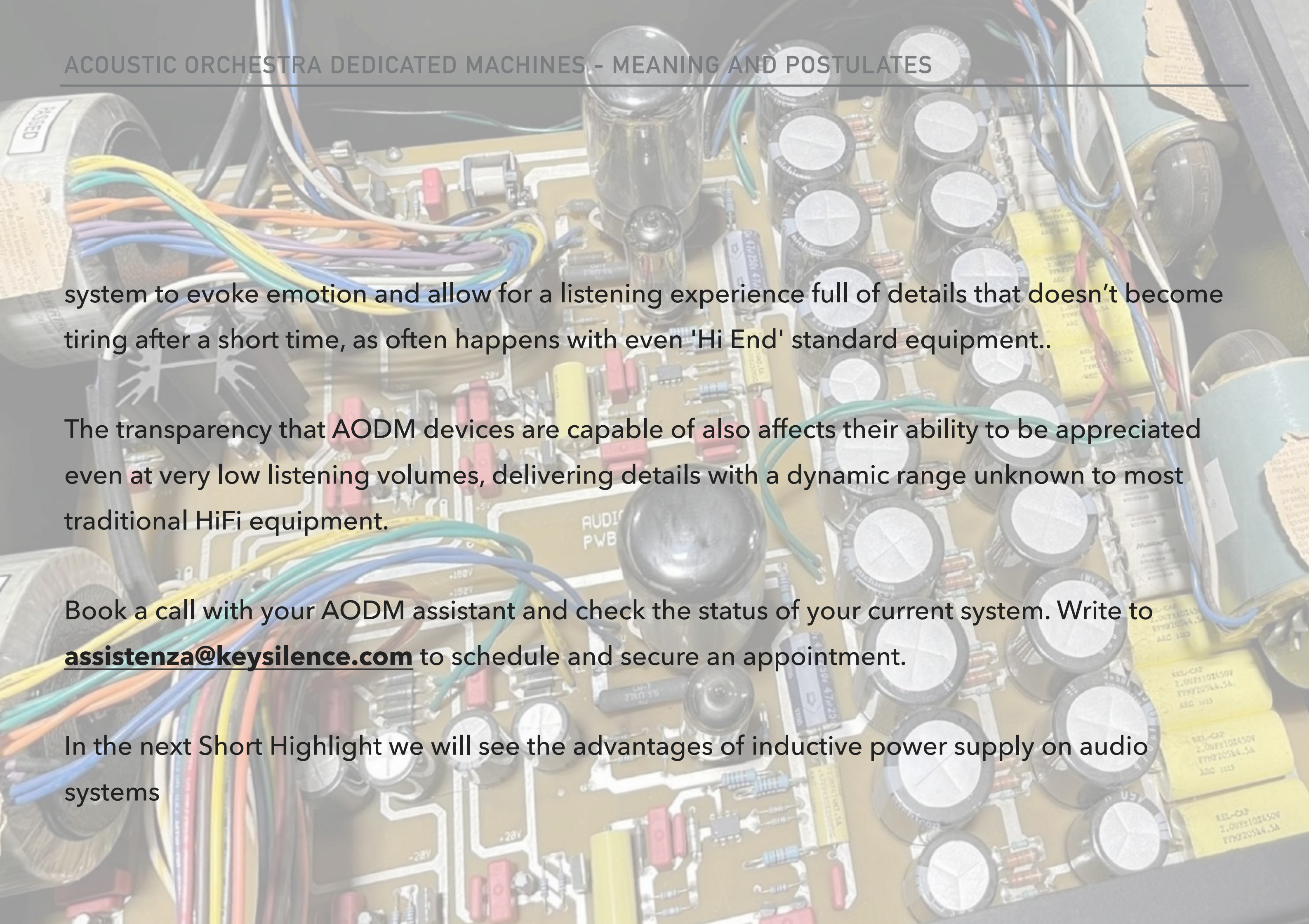
- Polyamide or polypropylene capacitors; still of excellent quality and included with high-end equipment

- Polycarbonate dielectric capacitors

- Polyester dielectric capacitors; these are the most commonly used

Given that the best capacitor is one that does not exist, there are specific AODM class 0 configurations without capacitors in the signal path, which are the most performant and transparent in handling the musical signal.

For us music enthusiasts, the AODM class represents proportionally the ability of a playback



system to evoke emotion and allow for a listening experience full of details that doesn't become tiring after a short time, as often happens with even 'Hi End' standard equipment..

The transparency that AODM devices are capable of also affects their ability to be appreciated even at very low listening volumes, delivering details with a dynamic range unknown to most traditional HiFi equipment.

Book a call with your AODM assistant and check the status of your current system. Write to **assistenza@keysilence.com** to schedule and secure an appointment.

In the next Short Highlight we will see the advantages of inductive power supply on audio systems

INDUCTIVE POWER SUPPLY

The quality of the power supply of audio devices, especially signal pre-amplifiers and power amplifiers is of fundamental importance, because the energy supplied by power supplies is actually what will turn into music, already!

The signal that will reach our ears is precisely the energy supplied by the power supplies after being "modulated" by the audio signal sent by the source, be it a streamer, a CD player or anything else that supplies a musical signal.

In traditional amplification systems (yes, even the most famous and/or defined as High END) so-called linear capacitive-type power supplies are commonly used, where the first "filter" stage after the conversion of the alternating mains current into direct current at 110V/230V, regardless of the country in which the appliance resides, it is indeed a capacitor.

The method is almost universally used because firstly it is extremely simple to size and, secondly, it is cheap and quick to create, not cumbersome and ultimately much lighter than inductive systems.

The performances of condenser filtering systems are often barely acceptable and do not guarantee any type of stability to the electrical current absorbed, in particular by power amplifiers, which in orchestral peaks can absorb currents of intensity far greater than that which the condenser system is capable of delivering.

Even in systems with multiple cascaded capacitors and large levels of total capacitance, limits on power reserves still tend to manifest themselves on the currents drawn by the higher frequency parts of the music programs.

Another of the most common problems of capacitor power systems, which in my opinion represents one of the major obstacles in the field of High Fidelity, is represented by the fact that even high "filtering" capacities are NOT able to block the disturbances of harmonic type present on the electrical network, and this, although it may seem surprising to the less experienced, is an important problem that affects the silence of audio equipment and their capacity for transparency and dynamics.

Inductive power supplies, on the other hand, unlike capacitive ones, are much more complex to size, cost significantly more, typically take up more space and above all are much heavier.

These "disadvantages", however, are all amply repaid by a much more stable energy that is purified from all the disturbances that can be found on the network at different times of the day. Harmonic disturbances generated by air conditioners, refrigerators, electric motors of all types and types normally present in our homes.

The result from a "sonic" point of view is simply NOT comparable and this is one of the essential reasons why inductive power supply has been taken as a reference model for High Quality equipment that intends to boast the AODM brand.

Book a call with your AODM assistant and check the status of your current system. Email **assistenza@keysilence.com** to schedule and obtain an appointment.

In the next Short Highlight we will see the advantages of systems without feedback between stages

NO FEEDBACK BETWEEN STAGES

The feedback between stages consists of taking a more or less small portion of signal at the output of an amplification stage to bring it back to the input of one or more stages that precede the sampling stage.

This circuit configuration, almost universally used, allows you to obtain very stable and "performing" circuits from the point of view of frequency response and above all distortion. In all those devices where the manufacturers / sellers are proud to show distortions in the order of 0.000... "something" %, rest assured, makes massive use of negative reaction or feedback.

If on the one hand, the use of the negative reaction offers notable circuit advantages and sometimes actually makes even tacky and improbable circuits made with components at the limit of decency usable in some way, on the other hand it has NEFASTIFUL effects on the audio signal and therefore on the program musical.

Whether small or large, the portion of the signal that is taken at the output of a stage and brought back to the input of one or more previous stages, being "in phase opposition" or negative in sign compared to the audio signal present at that point , actually disappears with the incoming signal, effectively canceling an important portion of its harmonic content.

The result? Many of the details in the music program will NEVER reach your auditory system. And its overall harmony will be heavily compromised.

When you listen to many of your favorite songs on an AODM system, you will most likely hear parts you have never heard, details you never knew existed.

The creation of circuits without feedback essentially implies a greater design effort and above all the use of more expensive technologies and components, which is why the vast majority of manufacturers go at random with the use of easily available and low-cost integrated components. And mind you, even within renowned products considered "high-end".

This is the reason, once again, why the AODM standard has excluded circuits that use negative feedback in favor of more expensive and better performing circuits without the use of tricks and shortcuts.

For the same reasons, the so-called “Operational amplifiers" integrated circuits were excluded, which by their nature must necessarily use the negative reaction between output and input.

In reality, as regards operational amplifiers, nowadays universally used also in the output stages of DACs, CD players on televisions and almost wherever it is necessary to pre-amplify audio signals, there are also other reasons that have led me to exclude them from maximum quality AODM systems, reasons which we will see in detail elsewhere and which in any case concern their internal complexity.

Email **assistenza@keysilence.com** to schedule and obtain an appointment.

In the next Short Highlight we will see the advantages of class A systems and pure symmetry

CLASS A ARCHITECTURE – PURE SYMMETRY

The circuits for amplifying a low frequency signal or an audio or musical signal, if you prefer, can be created in many different ways, with very different schemes and architectures. The principle behind a high quality device is that it is as linear as possible, i.e. that it introduces the lowest possible amount of distortion to the audio signal without altering its content and/or erasing parts.

A direct consequence of this principle is that the signal path must be as short as possible, that is, it must pass through the smallest possible number of components which obviously must be of the highest possible quality.

Despite this basic principle of physics and, I want to add, logic, in the world of audio equipment things have been seen (and still are seen) that a healthy mind would not dare to conceive.

Circuits that are incredibly complex or, conversely, simplified beyond belief to the point of omitting parts that, with good reason, should be considered indispensable.

But the thing that absolutely intrigues me the most (and arouses a certain level of hilarity) is how some technological solutions DESPITE their intrinsic defects, have been commonly accepted as traditional and therefore indiscriminately used as classical standards: complementary symmetry.

Complementary symmetry arises from the semiconductor industry's presumption of having "created" components which they say are symmetrical but capable of working with positive voltages and currents, while the other is capable of working with negative voltages and currents.

Now logically, by carefully observing the technical data sheets of these components, it is clear that these are symmetrical only on paper and that in reality, in the conditions of common use, these have significantly different behaviors from each other which it means that they are NOT mirrors of each other at all.

If for some uses these "symmetrical" components are extremely advantageous, the situation is different when they must work in pairs to amplify the audio signal.

On audio systems, yes, even those defined as Hi End, it is one of the main problems and presents limits for the uniform treatment of the signal which should translate into an exciting and relaxing listening experience at the same time, free from sharp edges and harmonic distortions which tend to tire the listening experience itself.

Complementary symmetry therefore treats the audio signal by dividing it into two parts: positive side and negative side. The two parts of the signal follow paths and pass through components that are NEVER perfectly identical - it's like having a painter decorate half of a room in your house with his "hand", his tools and his paints and the other half of the room to another professional with different tools and paints that will be ONLY similar to those used for the first half of the room.

Whatever the result, it will never be the optimal and perfect one as you wanted or imagined it.

This effectively asymmetric signal treatment afflicts the vast majority of audio systems.

Listening fatigue is even more pronounced when we use headphones rather than speaker systems.

For this reason, within the AODM postulates, the almost universally widespread complementary symmetry has been eliminated as an amplification solution and instead the more expensive and complex but also obviously more valuable pure symmetry or even the Single End configuration has been introduced.

While for the so-called Single End configuration which naturally operates in pure class A, the symmetric configurations, whether pure as per AODM dictates or complementary as in the vast majority of cases, the operating classes can be different from A and the most commonly used they are the AB classes and, in recent years, also the D class.

Class A (true) also represents a must for AODM systems and I will talk about it in detail elsewhere where we will also see the differences compared to other amplification classes that are so popular today.

For now I just want to tell you that class A for audio quality purposes is the best ever and to understand the reasons I'll give you a brief football example: in class A the electric current flows along the final stage continuously, it already has its own speed and so to speak kinetic energy. In class AB, however, the current is activated only when the musical signal requires it.

Now in class A it is like a footballer who, while running, reaches the ball and shoots it towards the goal, in class AB it is as if the footballer, while standing still, without the possibility of a run-up, has to kick the ball to send it towards the goal. Which of the two shots will be more powerful and dynamic? The conclusions are up to you...

ABSENCE OF CROSSOVER ON THE SPEAKER SYSTEMS

The problem related to capacitors and inductances as non-linear components present in crossover filters follows the same principle as capacitors and transformers on the signal path that we have already seen in the previous section dedicated to electronics.

Ideal conditions would suggest doing without it completely, however in certain circumstances it still makes sense to use it and optimize the distribution of frequencies to be reproduced on multiple speakers.

So what is the point of avoiding or limiting the use of capacitors and inductors on the AODM speaker system?

Before arriving at the answer to this question I want to make an important premise to keep in mind throughout the discussion. As we have already seen previously, capacitors and inductances are non-linear elements, that is, they change their behavior towards the signal that passes through them both as a function of frequency and as a function of current intensity.

Since the currents at play on the speaker systems are several orders of magnitude greater than the currents of the audio signal before it is amplified, it follows that in this context the negative effects that accompany their functionality are in fact more important than they they are on the low power electronic part.

Now let's get to the question. An AODM system is designed and therefore optimized for the optimal reproduction of acoustic music, i.e. made with acoustic instruments.

Among the sounds that an AODM system must be able to reproduce in an optimal and superior way compared to other systems, there is certainly the human voice. Since singing considers male voices among the lowest and female voices among the highest and taking into account sibilants and harmonics, we can safely say that this group of frequencies extends from just over 100 Hz up to over 5,000 Hz!

Reproducing this range of frequencies in a coherent and tonally correct way implies the need to use the same speaker over the entire range, avoiding like the plague interruptions and cross-over cuts to divide the range across multiple speakers, especially if they are different in terms of technology and dimensions.

These are the essential reasons why loudspeaker systems that aspire to bear the AODM mark must have no interruptions within this fundamental frequency range.

It matters little if the system has a sub woofer that works from 100 Hz downwards and a super tweeter that completes the emission after 12,000 Hz, these are aspects that concern a more general setting rather than a calibration for chamber music, the It is important to safeguard the integrity from 100 Hz up to 6,000 Hz.

So are AODM speakers absolutely the best in the world? The answer of course is it depends! It depends on the context. If you need to play Rock music, certainly not, there are more suitable speakers. But as far as acoustic music is concerned… the answer is certainly yes!

Acoustic Orchestra Dedicated Machines
Official LOGO

CERTIFIED
Genuine
AODM
CLASS 0
CERTIFIED

The audio devices which are in compliance with technical specs will show the AODM logo as above

DATE

SIGNATURE

Book a call with your AODM assistant NOW

and check the status of your current system

Email to assistenza@keysilence.com to schedule and obtain an appointment

a consultancy session on your system awaits you

and the amazing Qobuz Subscription

half price for one year (limited number - hurry!)

Until the day the laws of physics change, anyone who tells you “my devices are made without compromises” is just lying to you

Juan Del Vecchio

Author

- Of AODM Standards
- Of RHD technology
- Of AUDION SST technology
- Of the Juan ACT Method