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POWER (Part 1)

What output power should an amplifier have so that the loudness produced by an audio system is the same as in a concert hall when a symphony orchestra is playing at full strength?

How should the power of a loudspeaker system and the power of an amplifier be properly matched? How do the various types of power defined in domestic standards (rated, maximum, continuous, short-term, etc.) relate to those adopted abroad (musical, peak, RMS, etc.)?

These questions have always been in the field of attention of audiophiles. However, resolving them requires overcoming some mathematical difficulties. For those to whom the derivations presented here (mainly algebraic) seem too complicated, or who lack the patience to work through them, the main calculation results are summarized in tables.

The first part of this article is devoted to the basic concepts related to power, as well as to choosing the maximum sinusoidal output power of an amplifier that allows a predetermined loudness of music to be achieved in home conditions.

In the second part of the article, definitions of the various types of power used in audio equipment standards and advertising will be given, and the issue of safely matching amplifiers and loudspeakers in terms of power will be examined.

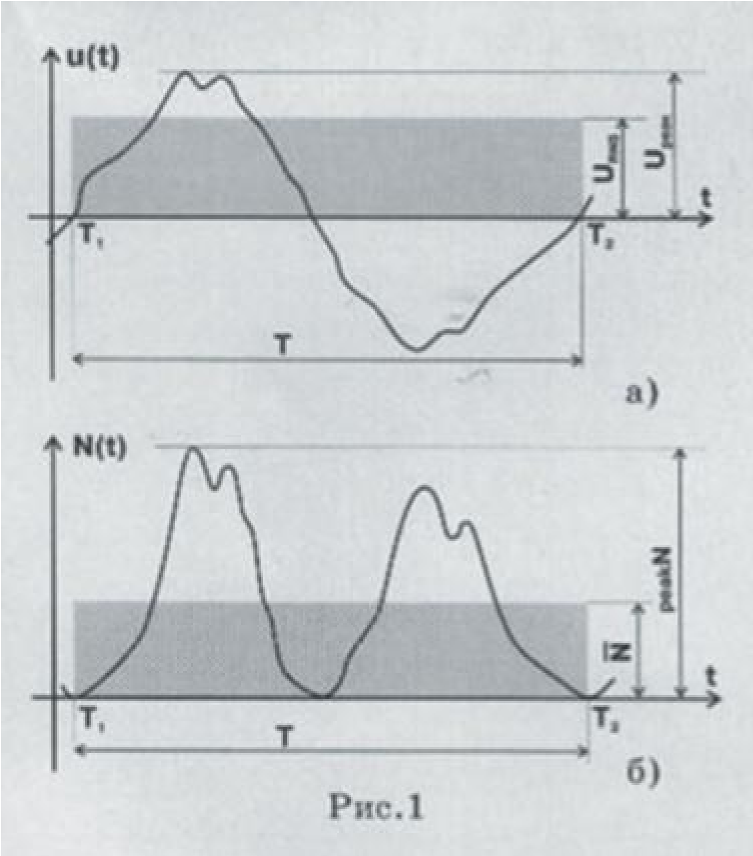
Usually we do not associate the word “work” with the pleasure of listening to music, nor do we calculate how many joules of energy an audio system “invests” into producing sound. At the same time, we are quite ready to discuss, with some expertise, the power of an amplifier or a loudspeaker. But do we really understand well enough what power actually is?

Let us begin with the fact that power, measured in such familiar units as watts, is actually a measure of work expressed as the number of joules of thermal energy released when that work is performed per unit of time (per second) $\Delta A(t)$.

A physics textbook gives a stricter definition: power is the limit of the ratio of the increment of work (in joules) to the increment of time Δt . It is written as:

$$N(t) = \lim_{\Delta t \to 0} \frac{\Delta A(t)}{\Delta t}, \text{ (Вт) (1)}$$

As can be seen, power is a function of time that represents instantaneous work, that is, work corresponding to any given moment in time. To distinguish N(t)N(t)N(t) from other concepts used in the article, we will call it **instantaneous power**.



You probably know that power can be acoustic, mechanical, electrical, etc. The law of conservation of energy allows us to express power of any type in the same units—watts—while adding a designation to the symbol NNN indicating the type (acoustic, electrical, etc.). According to the same law, power can be converted from one form to another—for example, from electrical to acoustic, as occurs in a loudspeaker. However, part of the useful power (usually a large part), contrary to our wishes, is "lost," meaning it is converted into heat.

To evaluate the efficiency of converting one type of power into another, the **efficiency coefficient (efficiency, η)** is used. It is equal to the ratio of useful output power to the power supplied to the input of the converter.

If useful power were not wasted and the efficiency of such a converter as a loudspeaker were close to 100%, then supplying it with just **1 watt** of electrical power would unleash the full force of a symphony orchestra upon the listener. However, real loudspeakers have an efficiency roughly comparable to that of a steam locomotive, and therefore amplifier output powers of **100 W or more** have become common among audiophiles.

So how much electrical power actually needs to be supplied to loudspeakers in order to achieve, at home, a loudness comparable to that of "live" music in a concert hall? Let us attempt to justify the choice of this power. To begin with, let us consider definitions related to the concept of the power of a musical signal.

If, during a musical performance, we observe with a dual-beam oscilloscope the instantaneous voltage at the loudspeaker input and simultaneously the instantaneous sound pressure produced by the loudspeaker and captured by a microphone, the following becomes apparent:

- the compared signals are very similar to each other;
- they represent oscillations;

- these oscillations have a complex form, not resembling a sinusoid;
- the shape of the oscillations, their amplitude, and their frequency composition change over time.

Since this article mainly concerns electrical power at the interface between the amplifier and loudspeakers, the question arises: **how can we move from the easily measured voltage at the amplifier output to the power delivered to the loudspeaker?**

This transition is made using the well-known formula:

$$N_{el}(t) = \frac{[u(t)]^2}{R_n}, \qquad (2)$$

where

- **N_{el}(t)** — instantaneous electrical power, W
- **u(t)** — instantaneous voltage at the amplifier output, V
- **R_n** — input electrical resistance (impedance) of the loudspeaker or its equivalent, Ω

It turns out that by recalculating instantaneous voltage, instantaneous power can be determined more accurately than by direct measurement using the reference method—a thermocouple. There are several reasons for this, but the main one is the thermocouple's inability, due to its thermal inertia, to track the instantaneous power values of a musical signal.

However, in most cases there is no need to know the countless instantaneous power values. Often it is sufficient to know only several characteristic quantities (see Fig. 1). Let us define them.

a) Average Power

Average power represents the result of averaging the instantaneous signal power over the interval between fixed moments T_1 and T_2:

$$\overline{N} = \frac{1}{(T_2 - T_1)} \int_{T_1}^{T_2} N(t)dt. \qquad (3)$$

Here and below, the bar above the symbol denotes averaging over time.

To express average power through the voltage at the amplifier output, the **root mean square (RMS)** value of the signal voltage is used. In foreign technical literature this quantity is denoted **RMS (Root Mean Square)**.

The RMS voltage is calculated as:

$$U_{RMS}=\sqrt{\frac{1}{(T_2-T_1)}\int_{T_1}^{T_2}[u(t)]^2\,dt} \qquad (4)$$

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- **u(t)** — instantaneous voltage, V.

If the RMS voltage is known, the average electrical power of a signal of any shape equals:

$$\overline{N_{эл}}=\frac{U^2_{RMS}}{R_н}. \qquad (5)$$

b) Peak Power

Peak power (peakN) is the maximum value of the instantaneous signal power within the time interval from T_1 to T_2:

max N(t) = peakN

c) Crest Factor

The **crest factor (F)** characterizes the ratio between the peak and average power of a signal within the time interval from T_1 to T_2:

$$F=\sqrt{\frac{peakN}{\overline{N}}}. \qquad (6)$$

For a sinusoidal voltage U(t) = Ua sin2πf, these quantities over the T2 –T1=1/f have the values:

$N_{el}=\frac{U_a^2}{2R_н}$; Urms = $\frac{U_a}{\sqrt{2}}$; Upeak = Ua ; F = $\sqrt{2}$.

If the musical signal did not change over time, the quantities we have introduced would be entirely sufficient. However, a musical signal is primarily characterized by its variability—that is, its dynamics. To reveal this dynamic nature, the concept of **dynamic power** is used.

Dynamic power is the **average power**, but unlike in formula (3), the averaging interval is not fixed; instead, it **slides along the time axis**. Therefore, dynamic power is a function of the current time and has the following form:

$$\overline{N}(t)=\frac{1}{\tau}\int_t^{t+\tau}N(t)\,dt, \qquad (7)$$

where **τ** is the sliding averaging interval of instantaneous power[1][1], measured in seconds.

The question arises: **what should the interval τ be in order to best reveal the dynamics of a musical signal?**

In sound recording equipment, two values of the constant **τ** have become widely used:

τ = 100 ms — this value provides the closest correspondence between dynamic power and the perceived changes in music loudness over time. This constant is used in sound level meters in the **"fast" mode**, as well as in the loudness indicators of mixing consoles, the so-called **VU meters**. The maximum value of dynamic power within a given time interval is denoted **max**.

τ = 2.5 ms — this value allows the observation of the maxima of slightly "smoothed" instantaneous power of the musical signal within a given time interval. This mode is used by sound engineers to determine how accurately these maxima fit within permissible limits. The slight "smoothing" ensures that exceeding the permissible signal level (in recording devices, amplifiers, etc.) corresponds to the listener's perception of distortion. The maximum instantaneous power measured in this way is called **quasi-peak power** and is denoted **peak**.

In modern digital recording level meters, both types of power expressed in dB are registered simultaneously: **dynamic power** (with averaging time constant **τ = 100 ms**) and **quasi-peak power** (averaged with **τ = 2.5 ms**). For convenience of observation, the latter is held on the indicator scale until a higher peak appears or for **2 seconds**, after which it resets, and so on.

It is important that the **instantaneous difference between the readings of the quasi-peak indicator and the dynamic power indicator** in the time interval between **T_1 and T_2** represents a quantity that characterizes the **crest factor** at moments of maximum dynamic power:

$$\Delta L_F = 10\lg\frac{peak\,\overline{N}}{\max\,\overline{\overline{N}}}. \quad (8)$$

The usefulness of this indicator will be explained later.

Nevertheless, our understanding of power would be incomplete if nothing were said about the **power spectrum**, that is, the distribution of the **average power of a musical signal across frequencies** in the standard audio range from **22.5 Hz to 22.5 kHz**. To distribute

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1[1] For simplicity of presentation, we consider moving averaging in formula (7) as occurring over a time "window," which can be represented as a rectangular weighting function of duration t. In measuring instruments, a moving exponential weighting function is used, which is implemented using a conventional RC integrating circuit. For this circuit, t = RC.

power across frequencies, filters with equal relative bandwidth are usually used—**octave or one-third-octave band filters**—and the average power is determined at their output in accordance with formula (3).

Recently, musical signals have also been described using a **three-dimensional representation of the power spectrum**, known as **FFT (Fast Fourier Transform)**. In this representation, a **time axis** is added to the power and frequency axes. Instead of a time-independent average power value in the filter band, the **dynamic power with averaging constant τ = 100 ms** is determined.

The definitions of **electrical power** we have given can also be used to describe **acoustic power**, except that instead of instantaneous voltage **u(t)** and electrical resistance **Rn**, we use **instantaneous sound pressure p(t)** and **acoustic resistance Ra**.

However, there are certain peculiarities in the use of acoustic quantities. Electrical power is transmitted through wires and therefore almost entirely reaches the load. Acoustic power radiated by musical instruments or loudspeakers in an audio system, on the other hand, **disperses throughout the listening room**, and only a small portion reaches the listener’s ears.

To determine the acoustic power acting within a specific region of space, the concept of **sound intensity** is used. This quantity is defined as the **ratio of the sound power flow in one direction through a surface perpendicular to the direction of sound propagation to the area of that surface**. The unit of intensity is **W/m²**.

According to the principles of **architectural acoustics**, the average sound intensity from a **non-directional source** (which may be a musical instrument or loudspeaker) at the listener’s position is calculated using the formula:

$$\overline{I} = \overline{N}_{ak}\left(\frac{1}{4\pi r^2} + \frac{4}{R}\right), \quad (9)$$

where:

- **r** — distance from the sound source to the listener (m)
- **R** — the **room constant**, which characterizes the ability of the walls, floor, and ceiling of the listening room to absorb sound (m²)

In this formula, the **first term** in parentheses represents the **direct radiation component** from the sound source, which decreases proportionally to the **square of the distance** from the source. The **second component** does not depend on this distance and is determined by **reverberation in the listening room**.

In any hall or listening room, it is always possible to find a distance from the sound source at which the two terms in parentheses in expression (9) become equal. This distance, usually called the **critical distance**, acoustically represents the boundary between the **direct sound field** and the **reverberant field**.

Later we will see that when listening to music in a concert hall or at home, we are located mainly in the **reverberant field**.

The critical distance can be determined from the formula:[2][2]

$$R_{crit} = 0{,}057\sqrt{\frac{V}{T_{60}}}, \qquad (10)$$

where:

- **V** — volume of the listening room (m³)
- $\mathbf{T_{60}}$ — the standard reverberation time of the room (s)

If you examine formula (9) carefully, you will notice that the average sound intensity at any point in the room (except in locations with possible acoustic anomalies) cannot be less than the average intensity of direct sound measured at the critical distance. Using formula (10), this conclusion can be expressed as:

$$I_{r>r_{zyn}} = \frac{\overline{N}_{ax}}{4\pi r_{zyn}^2} = 24{,}5\left(\frac{\overline{N}_{ax}T_{60}}{V}\right). \qquad (11)$$

However, sound intensity is not a very convenient quantity for practical use. Instead, the intensity level is used:

$$IL = 10\lg\left[\frac{\bar{I}}{I_0}\right], \qquad (12)$$

where $\mathbf{I_0}$ is the reference intensity $\mathbf{10^{-12}}$ **W/m²**.

The value of this reference intensity was chosen so that, under standard atmospheric conditions (ambient temperature **22°C** and static air pressure **750 mm Hg**) and under certain other conditions discussed below, **IL can be considered equal to SPL**.

And **SPL (Sound Pressure Level)**—the level of sound pressure—is a quantity familiar to everyone:

2[2] This formula is valid for omnidirectional sound sources. For sources with a figure-eight directional characteristic (e.g., an electrostatic loudspeaker), the echo radius value calculated from formula (10) should be increased by 1.7 times.

$$SPL=20\lg\left[\frac{p_{RMS}}{p_0}\right], \quad (dB) \qquad (13)$$

Where

p_{RMS} - RMS instantaneous sound pressure (Pa);

$p_0 = 2\times10^{-5}$ - the reference sound pressure (ISO standard threshold of hearing at **1000 Hz**)

For **stationary sounds** (i.e., those not changing over time), the transition from **IL to SPL** is valid when the intensity is determined according to formula (3).

To describe **musical sounds**, which change over time, we will use other quantities:

- **max SPL**, approximately equal to **max IL** when dynamic intensity is calculated using formula (7) with **τ = 100 ms**
- **peak SPL**, approximately equal to **peak IL** when dynamic intensity is calculated with **τ = 2.5 ms**

Do the physical quantities **IL** and **SPL**, and their variations, correspond to the **perceived loudness of music**? Psychophysical studies have shown that there is **no complete correspondence** between these quantities and perceived loudness.

Based on these studies, two quantities were proposed to describe **subjective loudness**.

The first is the **loudness level**, measured in **phons**. The number of phons equals the SPL of a **1000 Hz tone** that sounds equally loud as the sound being evaluated. Loudness level is directly related to **equal-loudness curves**. (AM” №4(5)95,c.63).

However, loudness level in phons is rarely used when evaluating music playback. Instead, the **A-weighted sound pressure level (dBA)** is commonly measured. This value closely corresponds to loudness level in phons because the measurement uses the **A-weighting curve**, whose frequency response resembles an **inverted equal-loudness curve**.

Nevertheless, **loudness level and sound pressure level in dBA** are quantities closer to the **physical definition of sound strength** than to the **actual sensation of loudness**.

To express **subjective loudness**, another quantity is used—**loudness**, measured in **sones**.

Loudness in sones reflects how the listener perceives the change in loudness when moving from one loudness level to another. The relationship between loudness level in phons (**P**) and loudness in sones (**S**) is expressed as:

$$S = 2^{\left(\frac{P-40}{10}\right)} \qquad (14)$$

Interestingly, according to this formula, relatively large changes in loudness level are perceived by listeners as relatively small changes. For example:

- A **10-phon increase** (a **10-fold increase in sound intensity**) is perceived as **twice as loud**.
- A **3-phon increase** (a **doubling of intensity**) is perceived as only a **23% increase in loudness**.

Let us now return to the physical measure of sound strength, **SPL**.

If you have followed the reasoning carefully, you can easily move from formula (11) to the relationship between the **sound pressure level produced by sound sources** and the **average acoustic power they radiate in a listening room**.

This relationship can be written in two forms:

$SPL_{r>r_{зук}} = 10\lg \overline{N}_{ак} + 109 - 20 \lg r_{гул}$, (15)

$SPL_{r>r_{зук}} = 10\lg \overline{N}_{ак} + 133{,}9 + 10 \lg T60 - 10 \lg V$. (16)

Formulas (15) and (16) can be considered correct when the distance from the sound source to the listener is greater than the reverberation radius. We will use these formulas, as well as the various concepts of power (intensity) we introduced, to systematize everything we know about musical sound levels in a concert hall.

What **maximum sound pressure level**[3[3]] can be expected when listening to a **large symphony orchestra**? [4[4]]

The data needed to answer this question were obtained by **Sevian, Dennis, and White (1931)**.

This information (see Table 1) is considered the most accurate, but unfortunately, it only represents the peak acoustic power emitted by the loudest musical instruments and a symphony orchestra. Therefore, using formula (16), this data had to be converted into peak

3[3] The values of max SPL of the musical signal calculated using formulas (15) and (16) can be considered correct when the reverberation establishment time in the room tyc=0.05T60<[t =100 ms].

4[4] Among natural musical sounds, a large symphony orchestra at fortissimo pitch produces the highest maximum sound pressure levels. For rock music, whose sound is produced using sound amplification equipment, characteristic sound pressure levels cannot be determined.

sound pressure levels (column 4 in Table 1) and then into maximum sound pressure levels (column 5 in the same table).

Table 1

Program Type	**Peak Power, W**	**Band of Maximum Power, Hz**	**Peak SPL, dB**	**Max SPL, dB**
Symphony orchestra (75 musicians)	13.8	250–2800	106	102
Orchestra (15 musicians)	9	250–2800	104	100
Organ (fortissimo)	12.6	20–62.5	105.6	102
Timpani	25	60–800	108.5	101
Bass drum	24.6	250–500	108.5	101
Cymbals	10	1000–16000	105	95
Cimbals / hammered dulcimer	9.5	8000–11300	104.3	95
Trombone	6.4	500–2800	102	97.5

These calculations were carried out for a **typical concert hall with a volume of 10,000 m³**, with a **standard reverberation time T_{60} = 1 s**, and for listening seats located **farther from the orchestra than the critical distance** (in our example, at a distance greater than **5.7 m**).

The values of the **maximum sound pressure level** were obtained by reducing the corresponding **peak level** by the value ΔL_f . This follows from formula (8). The values of ΔL_f were obtained by simultaneously observing the readings of a **quasi-peak indicator** and a **VU meter** while playing a large number of recordings of **symphonic and operatic music**.

The following values were obtained ΔL_f :

- for **tutti of a symphony orchestra**, as well as **woodwind instruments and the singing voice** — **3–4 dB**
- for **string instruments and brass tutti** — **4–5 dB**
- for **percussion and keyboard instruments** — **5–10 dB**
- for a **sinusoidal signal (as an example)** — **3 dB**

In subsequent calculations we will use ΔL_f = 10dB as the reference value.

The calculated **max SPL values** (after applying the correction corresponding to **A-weighting of the music spectrum**) agree well with the data of **Ioffe (1954)** regarding loudness levels of symphonic music corresponding to **dynamic markings in musical notation** (see Table 2).

To emphasize the data in Table 2, I supplemented it with values for the **probability of occurrence** of the indicated loudness levels in music and with the **loudness expressed in sones**.

It should be noted that the **most probable sound pressure level** at a symphonic concert is **70 dBA**, while the **maximum does not exceed 100 dBA**, and the sound pressure level corresponding to the **average intensity of a musical work** is approximately **77 dBA**.

Table 2

Dynamic marking	**Description**	**Loudness level (phons)**	**Loudness (sones)**	**Probability***
fff	Forte fortissimo (extremely loud)	100	64	2.5
ff	Fortissimo (very loud)	90	32	8
f	Forte (loud)	80	16	11
mf	Mezzo-forte (moderately loud)	70	8	35
mp	Mezzo-piano (moderately soft)	60	4	22
p	Piano (soft)	50	2	11
pp	Pianissimo (very soft)	40	1	8
ppp	Piano pianissimo (extremely soft)	30	0.5	2.5
* Values are approximate and depend on the genre of music performed.				

However, is it really necessary for **symphonic music reproduced at home** to sound as loud as it does at a **live concert**?

At what sound pressure levels do we prefer to listen, for example, to **forte fortissimo** at home?

A study of this question was conducted by the **BBC**. The results are shown in Table 3.

Table 3

Program type	Audience		Musicians	Studio Staff	
	Men	Woman		Men	Woman
Symphonic music	78	78	88	90	87
Light music	75	74	89	89	84
Dance music	75	73	89	89	83
Speech	71	71	84	84	77

It turned out that **listeners at home prefer much lower playback levels** than those heard in a concert hall.

Now that we know almost everything about the **actual and preferred sound levels of music**, we can conclude that for home reproduction it is sufficient to reproduce **max SPL = 100 dB** without distortion or overload for short periods.

In this case the **peak sound pressure levels** will not exceed:

- **108 dB** (in the band **250–500 Hz**)
- **106 dB** (in the band **500–2800 Hz**)
- **104 dB** (in the band **8–10 kHz**)

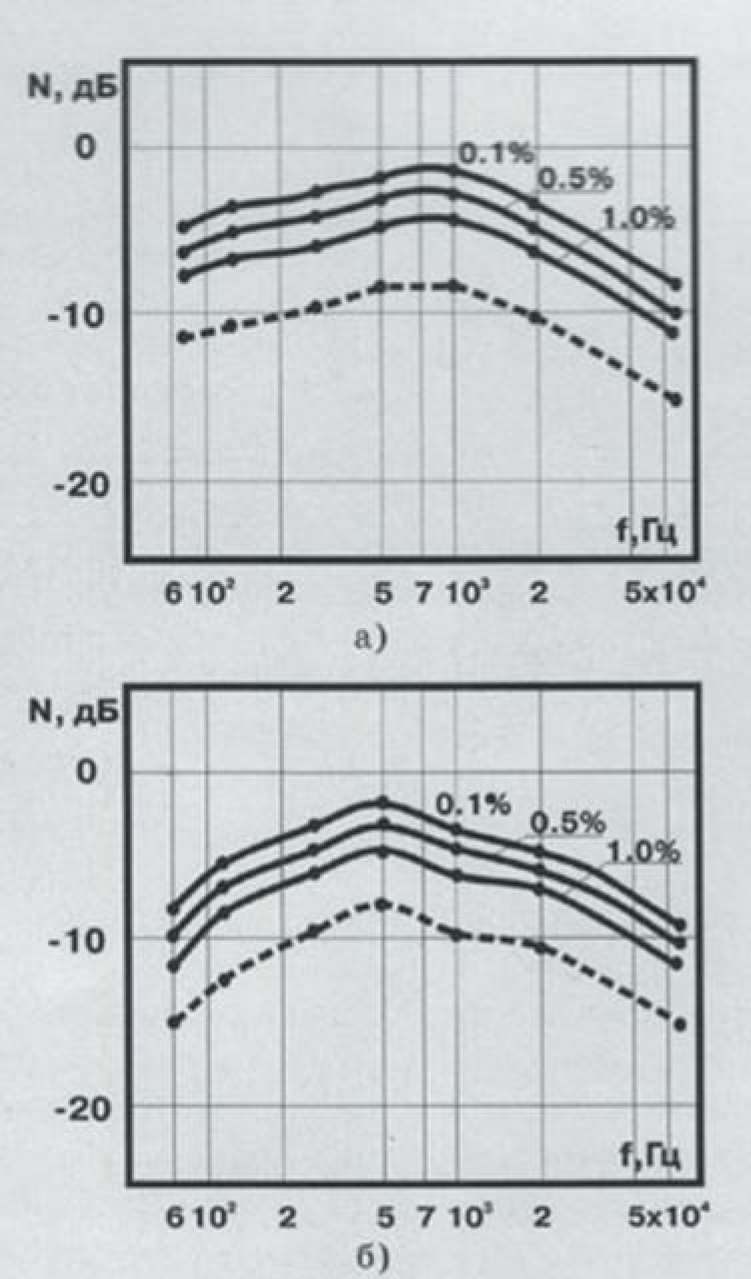

Рис. 2

Note that peak sound pressure levels **decrease above 500 Hz**.

This can be seen more clearly in measurements of the **power spectrum levels** (Shitov & Belkin, 1970), shown in **Fig. 2a for popular music** and **Fig. 2b for symphonic music**.

The **solid curves** show spectral levels of **maximum dynamic power** for different probabilities of exceedance.
The **dashed curve** shows the spectrum of **average power**.

From these figures it is evident that the **highest levels almost always occur around 500 Hz**.

How can an audio system produce sufficient sound pressure levels at home?

Let us return to **formula (15)** and attempt, using the **acoustic power of the sound sources** (in this case the **two loudspeakers**) and the **critical distance in the listening room**, to determine the SPL at the listener's position.

However, there is a complication: **how do we determine the acoustic power radiated by the loudspeakers?**

In principle it can be calculated by multiplying the **electrical input power** by the **efficiency**, but loudspeaker efficiency is **rarely given in specifications**.

Instead manufacturers provide **characteristic sensitivity (sensitivity)**.

Recall that **sensitivity** is the **sound pressure level in dB produced by the loudspeaker in an anechoic environment at 1 m distance on axis when 1 W of average electrical power is applied**.

Using formula (9), a relationship between **efficiency** and **sensitivity** can be derived:

$$10 \lg\eta = 10 \lg \; - 10 \lg \; = Sx - 109,$$

where

- **η** — loudspeaker efficiency
- **Sx** — loudspeaker sensitivity (dB/W/m)

This can be written as:

$$10 \lg \overline{N}_{ак} = Sx + 10 \lg \overline{N}_{эл} - 109. \qquad (17)$$

Now that we have the acoustic power of the loudspeaker, expressed through the characteristic sensitivity and the electrical power acting at its input, we will reduce formula (15) to the following form:

$$SPL_{r>r_{гул}} = 10 \lg \overline{N}_{эл} - 20 \lg r_{гул} + Sx. \quad (18)$$

If the amplifier operates **without overload**, then each value of **electrical power applied to the loudspeakers** corresponds[5][5] to a certain **SPL**.

This relationship ceases to hold when the amplifier output reaches its limit. When signal peaks exceed this limit they are **clipped**.

Clipping produces **nonlinear distortion**, perceived by listeners as **reduced dynamics, harshness, or scraping sounds**. It occurs because the amplifier **cannot deliver more power than it is designed for**.

Therefore the **maximum sinusoidal output power without audible distortion** that an amplifier can deliver to each loudspeaker is **the most important parameter of the amplifier**.

According to the international amplifier standard **IEC 268-3**, this parameter should be called:

distortion-limited output power.

However, not all countries follow this terminology, and many brochures and advertisements use different and sometimes contradictory terms such as **RMS power**.

We will clarify this confusion in **Part 2 of this article**.

Next we can use formula (18) to convert **amplifier output power** into **max SPL in the listening room**.

5[5] When determining the max SPL of a music signal, this statement is valid when 0.05T60 <[t=100 ms].

However, amplifier manufacturers use a small trick.

Amplifiers are tested with a **sine wave** (crest factor **3 dB**), whereas **music has a crest factor of about 10 dB**. To avoid distortion caused by clipping, the maximum dynamic power at the output of each amplifier channel should not exceed 1/5 of the distortion-limited output power. This can be taken into account by reducing the total distortion-limited output power of the amplifier channels by 7 dB when inserted into formula (18). Only then, by performing calculations using formula (18), will we obtain the maximum SPL of the undistorted musical signal that can be achieved with your audio system in a listening room with a known reverberation radius.

This radius, as it turns out, is almost always less than the distance from the listener to the speakers. For those audiophiles who might be tedious with the arithmetic, we present it in Table 1.

4 max SPL values calculated using formula (18) for different values of the amplifier's total output power, limited by distortion, with a known characteristic sensitivity of the loudspeakers, in a typical room with a standard reverberation time of T60 = 0.5 s [6[6]] and a volume of 60 m3 (which approximately corresponds to a room area of 20 m2).

For rooms of other sizes, an "addition" in dB to the max SPL indicated in Table 4 is introduced in Table 5.

Table 5

Room volume (m^3)	**30**	**45**	**60**	**90**	**135**
Approx. room area (m^2)	10	15	20	30	45
Critical distance (m)	0.44	0.54	0.62	0.76	0.93
Correction to max SPL (dB)	+3	+1.4	0	−2	−4

As an example, let's use Tables 4 and 5 to calculate the amplifier's required distortion-limited output power to achieve a max SPL of 100 dB with the characteristic sensitivity of the available loudspeakers, S = 87 dB/W/m, in a room with a volume of 90 m^3 (an area of 30 m^2). Substituting the room volume into Table 5, we determine the "addition" to the max SPL is -2 dB.

In Table 4, we select the diagonal corresponding to max SPL = 103 dB (with the expected value of max SPL = 101 dB "added"), after which we connect the characteristic sensitivity

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[6[6]] In IEC publication 543, the standard reverberation time T60 = 0.5 s is recognized as typical for residential environments.

value of 87 dB/W/m with the diagonal of max SPL = 103 dB and then with the required power, as shown in Table 4 using the arrow. The required output power, limited by distortion, must be at least 40 W per channel.