AUDIO MASTERING

IN DIGITAL ENVIRONMENT

For beginners and intermediate

Ferenc Gyuricza

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PREFACE

When I came up with the idea of writing a book on the diverse subject matter of mastering, I was aware from the very beginning that this was an impossible task, as it is extremely difficult to write a book about an audio technology process which mainly requires the use of our ears. This is the reason why most books on mixing and sound technology do not dare to take a deep dive into the subject, and I want to avoid superficial meddling. As the topic is very diverse, I had to delimit the content of this book. This is why I decided that the book will solely be about the presentation of digital mastering processes, with a deeper analysis of the effects. Many people use self-education to begin their journey on this technology, since they do not have the required funds to hire a professional, or they would like to learn the nooks and crannies of the field by themselves. So this book is primarily meant for them in an easily comprehensible manner, but I hope it will provide a lot of useful information that can help skilled audio engineers as well.

WHAT IS MASTERING?

Mastering refers to the process of improving and refining audio material. It is used mainly in post-production of mixed songs, in the final phase, where possible noises are eliminated, spatial perception is modified, the overall picture is modified via EQ, the volume is increased and decreased, as well as certain frequencies are suppressed or amplified. The aim is to have the songs on the disc sound in a unified way, thus providing the listener a pleasant effect. Any software or hardware, which modifies the original source in any way, will assist in the mastering process.

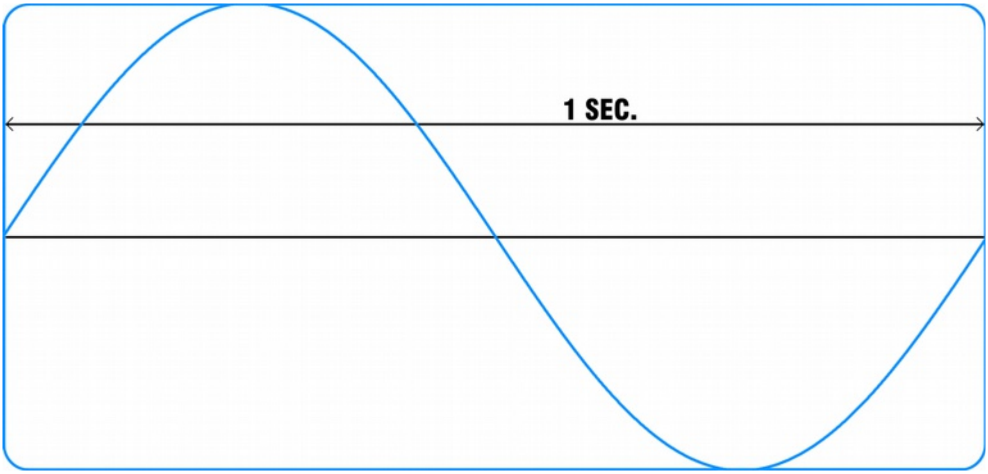
Today, the term mastering became quite chaotic regarding its exact meaning. It was easy a while ago. There was a recorded song that had to be improved or adapted to the demands of the given age. Today, most people are already using this feature on DAW (Digital Audio Workstation), the master channel of the music editing program, during the phases of mixing. Some people consider mastering as follows: the song is mixed in such a way that there is no signal-shaping effect on the master channel other than the limiter. But I would like to emphasize that this is not mastering in the strict sense, it is mixing. The latter solution can be as effective as traditional mastering. However, most people use the two procedures together. That is, the tracks of the music editor are mixed and the master plug-in's settings on the master channel are changed in real-time to achieve the desired effect. Less experienced musicians believe that mastering is used to achieve the greatest volume. The latter phenomenon is, by the way, a terrible crazy thing, since we want to use it even when it is not needed at all.

Most of the mastering processes that are commonplace in today's world only require a computer. In the current modern trend, all available techniques can be used for mastering, if these are to the advantage of the mixed material. According to the classic concept, mastering, on the other hand, applies only a finishing touch to the final material, which does not require a complicated toolbox, just quality tools. I do not aim to argue with anyone about which one is better, because I believe that both approaches are great if the circumstances are right.

I understand the squeamish because a $200 software is far more affordable than a $10,000 Fairchild compressor. That is why everybody can use it, which results in a lot of failed materials reaching the market. The exclusive nature of the profession is beginning to disappear. But the software gives you the same opportunity for doing something the right and the wrong way as well. This is not the developers' fault. Furthermore, when viewed under a creative light, today's software offers a much larger scope for creativity than the use of best practices. I believe that this is a great benefit for those who create content in our age.

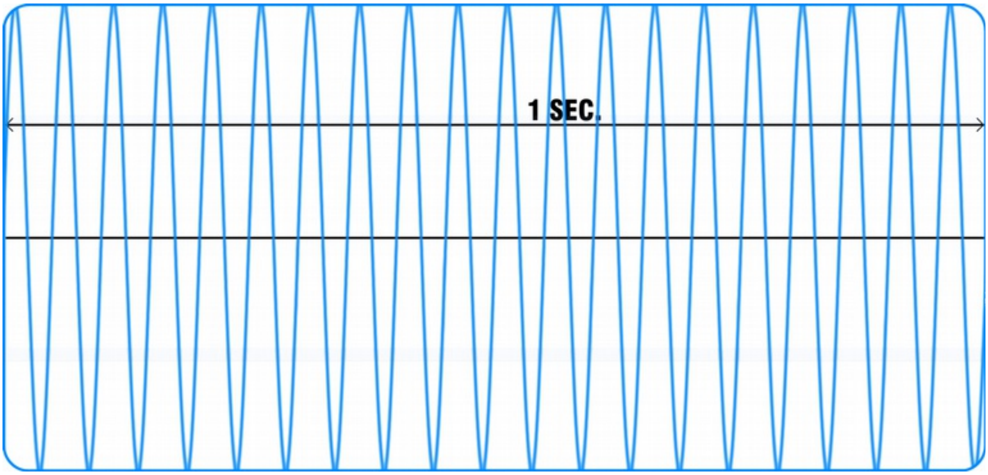
However, in order to be able to talk about mastering effectively, I have to turn to an aspect of creating sound which is seemingly boring, yet pivotal in truth.

WHAT IS CONSIDERED SOUND IN THE DIGITAL FIELD?



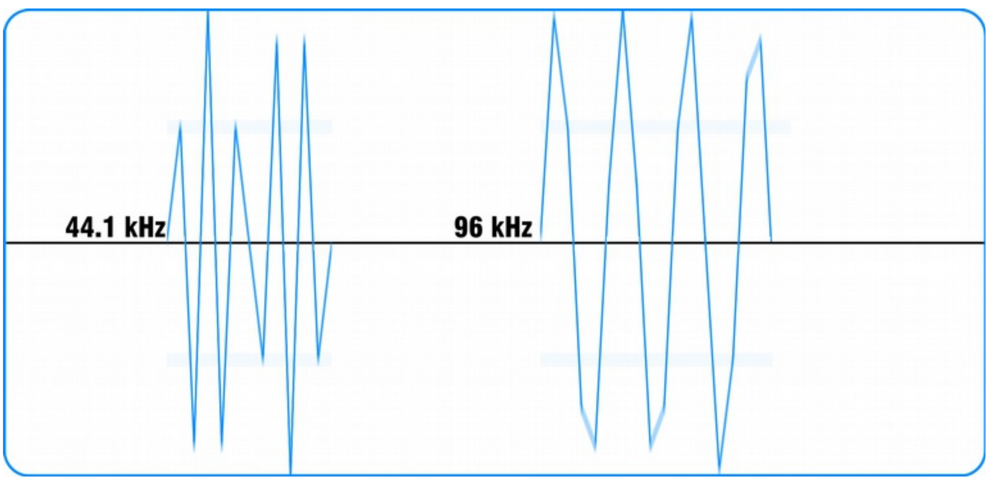
Analog audio, which lacks any overtones, is a sound wave. In general, sound waves perceptible by human ears are between 20Hz and 20kHz and these are represented by sine curves. In the above graph, 1Hz indicates an exactly 1 second-long sound wave. If you see 20 sine waves within 1 second, this will produce the sound with the lowest vibration a human ear can hear. Not all studio monitors can sound this pitch. In many cases, it is not even necessary, but sounding above 30Hz is already an important requirement as this pitch is already clearly audible in any recording.

If we see 20,000 such waves in a second, that is the highest sound vibration we can still hear. Of course there are differences, everyone’s ears have different sensitivities, but generally, this value is considered true. The capabilities of all audio equipment are adjusted to this range. So the sensitivity of your microphone and the capability of your studio monitor are calibrated to these or wider ranges.



The sinusoid is not found in nature, and analog sinus generators cannot perfect the sinusoid either. In fact, every existing sound ever made can be produced by mixing different sinusoids. If you mix two sinusoids of different frequencies, you have mixed a new tone! When you speak, your voice is unique, since the voice cords do not create a single tone. Almost every sound has unique vibration, overtone, and harmonic vibration. The overtone is a good example in case of an acoustic guitar. If you pluck the cords of an acoustic guitar, let's say the c sound, the guitar will sound an octave up, and if you have great ears, you will hear it. Very softly, but it is also in the guitar’s sound. The harmonic overtone that is a pitch is emitted by a cord or column of air when plucked at its half, one third, one fourth, etc. If you like, keynote can be sounded with an octave, 7 semitones, etc. higher. This means that their frequency and volume provide the unique sound. That's why your voice will be different from everyone else’s. From this point on, it is clear that an analog sound is actually made up of an infinite combination of sinusoids. It is impossible to model this incredible complexity in the digital realm. That is why analog synthesizers sound better, or if not better, in more detail. There is a reason for people saying that there is a huge difference between analog synth and a digital plug-in!

So why are we bound by the limits of digital technology? The birth of the compact disks heralded a technological leap, but the process had to be regulated by standard rules. We had to include an entire album on a disk, and we needed to produce the best possible sound quality with the digital tools of the era while avoiding overburdening our hardware, when an album was made or when the CD player was playing music. This is why the music was recorded on the CD with a bit depth of 16 bit and a sampling rate of 44,100 signals per second. This became the standard, which enabled us to listen to high-quality music. Today we consider 16 bit and 44.1kHz to be a joke. In any case, this sound quality is a significant barrier in working with the best quality at home.



I will show you what this is about. We have a song, which is made with a sampling rate of 44,100 per second. Let us suppose, probably correctly, that our music will include the highest frequencies detected by the human ear, namely the 20kHz range in this case. With the use of the Sound Forge software, I generated a sinusoid of 20kHz, which can be observed at a sampling rate of 44,100 per second. This is what it looks like:

Where did the nice even curvature of the lower frequencies go? At the highest audible frequencies, the richness of detail seems to disappear because of the inability to render 20,000 sinusoids in 44100 signals for these to appear rich in detail. This will make some sampling points quieter, some louder, and it will look like a sawtooth signal. However, according to Shannon’s sampling law, a continuous signal can be perfectly recovered from its samples if the sampling frequency is more than two times larger than the signal’s bandwidth. What does this mean? That if the audio is sampled at 44.1kHz, it will sound exactly like a 48kHz audio in the 20Hz to 20kHz range. In practice, the D-A converter of the soundcard produces a continuous signal from the fractured one. If it does its job badly, the difference is audible, if it does it well, this cannot be heard. This is typically an output quality deterioration, that is, the quality of the soundcard’s output audio deteriorates, not the quality of the audio itself. Another example of quality degradation is when the DAW runs at a different sampling rate than the audio used. In this case, the DAW adapts the audio to its own values, which can lead to a quality change as this is an internal process. I would like to add that this is not present in every DAW as a problem.

What is the solution if we use several loops and most of these are recorded in 44.1? We can work with it, but the DAW should always be used above 44.1kHz! Obviously, if your computer and soundcard can take care of it, then at least 48kHz is strongly recommended. Why? Although our sound sample is recorded at

44.1, if we apply effects to it, the effects will work at the values specified by the DAW. Or, if the DAW works with higher values, it will interpolate each sound sample to its own value before it can be heard. The raw signal alone will not be richer in detail, but you can already hear the difference when applying effects. At this point, many people ask themselves, what is the point of working at a higher sampling frequency if the music is downgraded to 44.1 anyway? As I mentioned earlier, the analog audio signal consists of an infinite number of combinations, and we strive to approach this effect. The music's richness in detail will be different if we work at a higher or lower value. Even if we downgrade a higher quality music, the end result will be better after the downgrade than after the basic bit rate, because the sound will no longer depend on the sound quality of the DAW's output, but on how we recreate our music richer in detail in 44.100 sampling rate. The difference is seemingly negligible, but since there are so many seemingly negligible moments in music editing and mastering, it all depends on the effective sound of music.

Higher sampling rates are used for movies and DVDs. It is better to adjust to this, and if for some reason 44.1kHz material is needed, convert the end product with a great converter. A good converter is truly important here. In order to be able to edit music, it is essential to use music editing software. Whichever software we choose, the sound quality is important. Many believe that there is no difference in sound quality between programs, but that is not true. Every software uses a unique programming solution, and when we edit multiple tracks in the music editor, there are processes in the background that we do not even think about. One of these is managing the sample conversion.

Our digitally recorded music requires a minimum of 44,100 recognized signals per second. The highest audible frequency is

20,000Hz, so a computer must record more than 40,000 signals per second for the waveform to appear in the recorded material. However, the music editing software must also handle audio samples recorded at different sampling frequencies. When an audio sample is stretched or a different sampling frequency is played, the software must calculate how the audio track will sound at that sampling frequency. This is done to follow the current frequency of the soundcard. Software do this with varying degrees of accuracy, so there is a music editing program that does its job better, and others do it worse. This means that some music editing programs may be scolded for their sound quality, while others may claim that a particular DAW produces perfect sound quality. In terms of resampling, Ableton 9.11, Logic X and Pro Tools 10 rank among the best, while Cubase and Reason have the worst results. Anyone who has used resampling in Sound Forge in a PC environment should choose other software for this task as it carries out pretty ugly changes.

Bit depth is the next factor. This is more important than the sampling rate. If you have set your soundcard or DAW to 16-bit before, switch it to 24-bit! Music data stored in 16-bit can take 65,536 different values, while in case of 24-bit, tone date is calculated based on 16,777,216 different values. 1 bit is 6dB, which means that the whole dynamic range is 96dB for a 16-bit material and 144dB for a 24-bit material. This is a significant difference. It is less noticeable in terms of sound quality, but even a little difference is noticeable in the end result. So, if you turn down a 16-bit audio track to -96dB, nothing is left of the audio track. With 24-bit, there is still useful data in your track. Obviously, no one works with -96dB audio tracks, but now we know that bit depth affects the dynamics, and the volume data is stored in these bits. As for the mastering process, let's say you get audio for 16-bit mastering, which is extremely hollow because it has little high-frequency range, and whoever exported the material to a stereo track, the track was not rendered at maximum volume, which means that the signal peak will only reach -6dB. In this case, the material has already lost 6dB from the 96dB useful dynamics. As the music is hollow, we need to raise the volume of the high range using an equalizer. At worst, this can be extremely high in dB. Nevertheless, let us suppose we raised 6 decibels in the high-frequency range when using EQ. and even had to raise the low-frequency range. So far, we have lost 12dB. Then we use a widener as the stereo space of the music is too narrow. With a widener, we can separately handle signals on the edge and mono signals. Increasing the volume of the signal at the edge only increases the space. In our particular example, we increase the space by 4dB. Then, after doing all of this, we put a maximizer on the music with a threshold of -6dB, which will make the music louder. In this example, 22dB was lost and the noise level of the audio increased to -74dB. The noise itself is not heard in the music, but the music will have a more distorted effect for some reason. This can cause problems, among other things.

I do not claim that noise is always a problem, as old tape recordings are noisy as well. All I say is that these do not always meet today's requirements. An airy music with less instruments has fewer problems than a very dense instrumental music. If you no longer use 16-bit mixing, in many cases the mastering of your song may end up better. For example, if you use dense reverb on an instrument, reverb can also produce more complex reverberation when using 24-bit. The other extreme is setting the DAW to 32-bit 96kHz. This is superfluous because hearing the difference is not due to the quality of the audio sample, but to the processing capabilities of our soundcard.

STUDIO MONITOR

This is a very important tool for mixing and mastering. The difference between a traditional speaker and a studio monitor intended for a studio is no longer negligible. If you have been mixing on Hi-Fi speakers in the past, drop this habit because it will take you in the wrong direction! Even with more affordable studio monitors, you are better off than with an average speaker. Why? These special speakers are designed for this purpose.

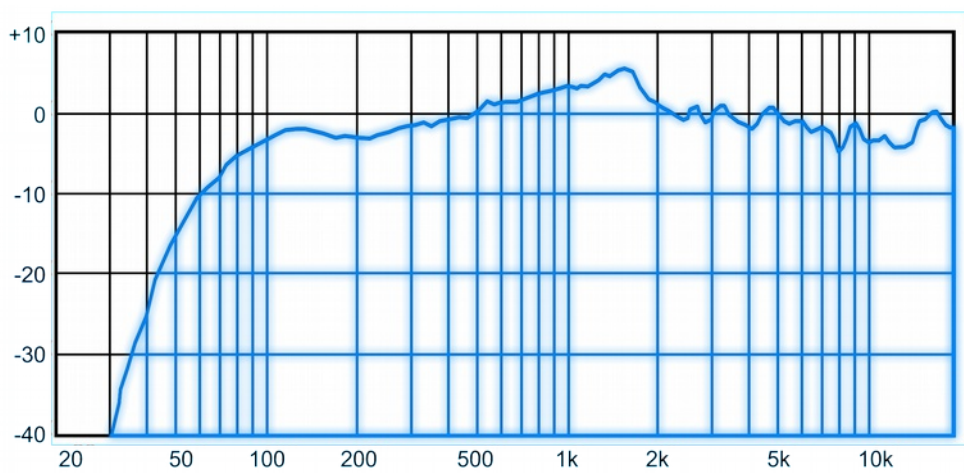
Many speakers have crossovers. What is this? Basically a filter. It allows only the low frequency into your low speaker and only the high frequency into your high speaker. But things that can be described as simple can become very complicated. Every speaker has different solutions, and not all of these are good. Even the shape, size and material of the box matter. It is obvious that a plastic box has worse performance than one made of wood. Most Hi-Fi speakers have electronics that alter the signal in terms of sound fidelity in a disadvantageous manner. By Hi-Fi speaker, I mean cheap mass-produced speakers. However, when developing a studio monitor, care is taken for the sound, electronics, speaker, material and size to be in harmony. Obviously, this may not perfect in the case of every studio monitor, but there is also a significant difference between a Hi-Fi and a cheaper studio monitor. Whoever switches to a studio monitor will not be able to mix well at first. Some take weeks to adjust, others get used to the new sound in a few days. In terms of Hi-Fi speakers, only the expensive ones can be considered an exception, but these are more difficult to choose from, so if you can, stick to the device which is developed for this reason!

There are a few other things to consider when purchasing a studio monitor. One aspect is the size. If a smaller studio monitor that can sound bass like a large system, we should be skeptical. In this case, the sub-range is not the result of the box's size, but the fact that electronics are built-in that enhances low frequencies. The disadvantage of this is that the subwoofer receives more load. In this case, the speaker must be designed by the manufacturer to accommodate this extra work. However, the more electronic fraud, the more the signal changes before it reaches the speakers. Make no mistake, these studio monitors can be good, but it is very important that we test the sound of the device before the shopping fever takes over our mind. These devices are excellent as control monitors, and even if you do not spend a lot of money on sound, these stand their ground. In general, the price more or less reflects performance.

In the case of studio monitors, some manufacturers are labeling cheap speakers as studio monitors in the hope of profit, which can be considered misleading the customer. Typically, such a device is the Fluid Audio C5, which is sold as a studio monitor, but the sound quality is unacceptable and sub-par. The low frequency in this small box almost rips out the diaphragm even at medium volume, but at the same time results in the tremendous distortion of the signal. The elaboration and the stereo space is poor. So the point is that avoid very cheap products and, if possible, only order a monitor from the internet without testing it, if it comes from a reputable manufacturer.

Another important factor when purchasing a monitor is linearity and frequency transfer. As mentioned earlier, the human ear can detect sounds between 20 and 20,000Hz. A monitor that is not capable of emit sound below 75Hz will be a major drawback when mixing. Let us pay attention to this and consider the aspects! Most manufacturers release a unique sound specification for their monitors, that is, the speakers are measured and this value is unique to each piece. When purchasing, it is worth paying attention to the extent to which each speaker is capable of transmitting a specific frequency range. Of course, a monitor with maximum capacity is not always required. For example, if you are editing videos, you may not need to play extra low frequencies. If music is only your hobby and you do not want to consider this as your work, then a cheap, smaller monitor with a good headphone is more than enough.

Why is linearity important? The answer is simple. If a monitor emits a frequency range louder, it is likely that during mixing or mastering, this will trick you and you will lower that range because your ears consider linearity natural. After a while, you will get used to it and learn to mix on it properly, but for the main listener, you should choose a monitor with less oscillation! If this is true, why was the old fossil, the Yamaha NS10, so popular in studios? Not for its beautiful sound, I can guarantee that. The NS10 always functioned as a control, secondary interceptor in reputable studios, as its sound was so "low-key" that if there was an enormous mixing mistake, it could be heard immediately on this speaker. Long term mixing is not good on such a speaker because it will wear out your ears, but it is great for pointing out mistakes.



Yamaha NS10 frequency curve

The production of the Yamaha NS 10 speakers began in 1978 and the last one was made in 2001. These are 2-way speakers with low/middle and high-pitched functions. These speakers are still passive, i.e. the signal is amplified by a separate amplifier. Over the years, it has been named the standard speaker in the music industry. However, the reason for this is not that it sounds good. If you are thinking about purchasing such a speaker system, keep the following things in mind. The sound of this speaker is not perfect at all. But if it is not perfect, what is good about it? By default, if you mix music on a state-of-the-art linear active studio monitor, depending on your knowledge of the capabilities of our speakers and what we are capable of, we will most likely mix well. However, a small mistake may slip into the mix and this is when a secondary monitor comes in handy. A secondary monitor is always recommended to be a speaker which has completely different characteristics than what you use for mixing. This is where the Yamaha NS 10 is useful, as its sound is non-linear. The accuracy of a good studio speaker depends on a number of factors, one important element of which is the sound output at a constant volume throughout the audible frequency range. In white noise all frequencies are present, and these have equal amplitude and volume. If the speaker is capable of delivering white noise so that the sound output from the speakers displays the amplitude of each frequency with almost the same peak volume, then this is a linear speaker and, as a result, the speaker will reproduce the frequencies of the music to be mixed in a perfect manner. The Yamaha, however, differs from this linearity. At 2kHz and above 10kHz, it sounds much more intense and makes the lower frequencies sound hollower. This is why it is basically not good for mixing music on this speaker alone. Only professionals who are already very familiar with the capabilities of the speaker can do this. Of course, it is okay to mix music only on this, but if you do not know the capabilities of the speaker, it is almost impossible to make a good mix using only the Yamaha NS 10. However, it is a great choice for a secondary monitor, precisely because its sound differs from linear. If you raise a tone too much in the mid-frequencies in the mix, it will immediately be heard on the Yamaha, as the mid-range will automatically be louder. The same is true for high frequency. If we check this for incorrectly mixed music, the sound will damage our ears. The point of this speaker is not to sound good in all circumstances, but to reveal the flaws. Another great advantage of the Yamaha NS 10 is its small woofer size. The smaller a speaker is, the more likely it is to display transients, bouncing sounds, or sound waves richer that are suddenly changing in the signal stream. A speaker with a large-diaphragm finds it is difficult to handle these sudden changes. Obviously, the tweeter transmits the transients pretty well, but the sub/mid-woofer will no longer handle sound with suddenly changing intensity. Due to the size of the diaphragm, Yamaha excels at displaying these sudden changes. Not only the size helps this effect, but it also renders the sub-bass less visible, making it possible to reproduce other frequency ranges more accurately without masking the sounds of the sub-bass ranges. Older types of speakers are closed, thus lacking a reflex port. Among other things, this is why it has fewer low frequencies, but this is the reason for its clearer sound as well. Since there are quite a few subtypes, you should be aware of the differences.

NS10M is the oldest. If you want to buy one, take a closer look at it because it may not be in good shape due to its old age, but its sound may be better than later versions.

The NS10M Pro, NS10M Studio and NS10 MC are newer models with higher performance, and these can sound lower frequencies even louder.

The NS 10MX and MT are the latest devices. The production of MX began in 1993 and the MT in 1996. The latter also has a reflex port in the front. These types are already magnetically shielded as well.

There is another model which may be suitable, the Yamaha NS 615. The components of this model are identical to the electronics of the NS 10. For some reason, the price tag is a fraction of the Yamaha NS 10.

These speakers also have their downsides. As they are passive, a good amplifier is a must. If it is not connected to a good quality amplifier, it will have greatly reduced sound quality. If you are a perfectionist, the NS 10 is not a cheap purchase at all. The further downside is that these products are used and old. Although there is room for error.

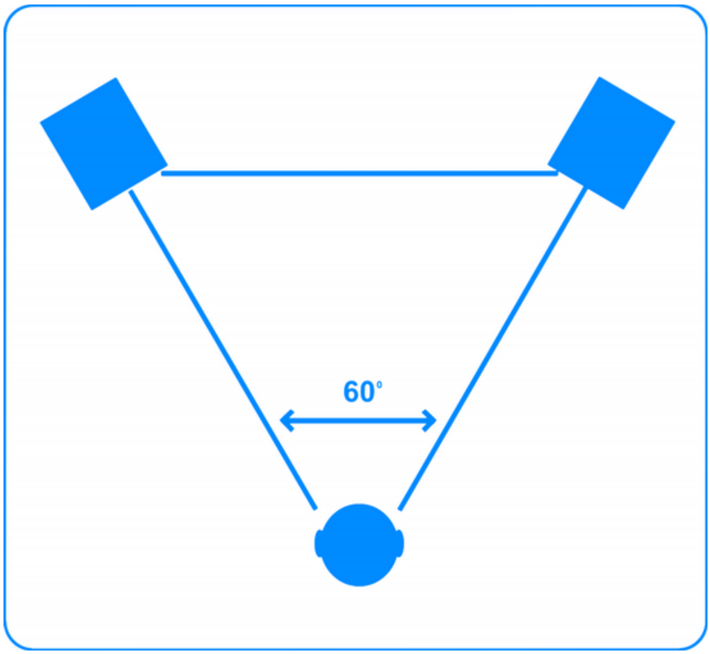
What speaker can be considered if the Yamaha NS 10 is not an option for us and we want something more modern? Well, the active speaker similar to the capabilities of the NS 10 is the studio monitor Yamaha HS50M. This is almost the same for the NS10 in terms of frequency peaks. As it is active, there is no need for an amplifier, and we can also work with low, mid and high ranges on the backside. Unfortunately, this speaker is no longer manufactured, only used models can be purchased. Its advantage is that it is smaller and not so old that we have to be afraid about it becoming ruined. If you do not want to bother with this, I recommend the Yamaha HS5 because it is still available for purchase, with the restriction that the sound of the HS5 is no longer so similar to that of the Yamaha NS 10, but it is still a much better choice for a secondary monitor than a more linear speaker. I have to say that I personally find the sound of the old NS10 better than the alternatives listed above.

If you want to get two separate studio monitors, be sure to choose speakers with different characteristics! You do not have to purchase the NS10, it is a waste of money in a smaller studio. Some people work with three different speakers. They compose music on one because they like its sound, the other is used for mixing, since it is linear, and use the third for testing and finding errors. If you do not have the required funds, you can test your almost finished music on a basic Hi-Fi system as well. The aim during mixing and mastering is making your music sound good on as many systems as possible. An everyday listener does not have an expensive Hi-Fi speaker at home, so you need to think about the average music listeners as well. Most people do not care about testing, as it goes online anyway. Unfortunately, this is a general phenomenon and does not help the musical taste at all.

A studio monitor has a passive and active speaker. Active speakers already include an amplifier, so you do not need to buy an amplifier separately. Just plug it into the audio interface output, provide power and you are done. Passive speakers obviously need an amplifier to drive the system.

Some monitors, almost all, have a function where frequency control can be used to correct the unfavorable sound of a studio room. If you are working in a studio room with mediocre soundproof walls and the sound is unfavorable, or you may not be able to position the speaker properly, this feature may be useful for correction.

When purchasing a studio monitor, be aware that there may be differences in capabilities between the same brand and types, so look at the frequency distribution and test the speakers! If you find that your monitor is capable of delivering the correct low frequencies but does not do so, or there are too many low ranges, try moving the monitor to another position! For example, further away from the wall.



The ideal setup is as follows: So the distance between the two speakers and your head is the same, and the speakers are facing your ears.

A small room can have a detrimental effect on the sound, but it helps a lot when you put dampers on the walls of the room. This is not a soundproof material, but by its name, it dampens reverberations from the wall. There is no need to cover the entire area of the walls because the goal is to have the right sound wherever you sit.

Although it cannot be seen on the image, it may be worthwhile to place a dampener above the head on the ceiling.

Of course, it is also possible to cover the entire wall surface, but the acoustics of the room will not end there. It is worth placing bass traps at the corners of the walls. There are many ways to correct low frequencies, so I would not delve into this topic any further, but the basic concept is to place them in every corner of the room. Use it behind the speaker is especially recommended. In many professional studios, monitors are built into the walls, thus improving the sound of the room. Speakers installed close in the room are called near-field monitors. These are typically small studio monitors and are great for smaller rooms. It is worth placing the larger devices away from our ears.

It is important to highlight the clear fact that the main purpose of monitors is not to always have great sound. The purpose of the system is to hear all frequency ranges properly, with its fine and faulty parts, to have excellent sound space, and to hear transients well. Most Hi-Fi speakers are not suitable for this. This is why linearity is extremely important. Whose ears are accustomed to the sound of a less linear piece can mix or master, but professional users cannot afford to deviate from it as this is the standard they always cling to. The reason for this is that professional studios can tell you why a speaker’s characteristics are not good, while many who have not listened to professional equipment can attribute the “great” label to an average speaker. An important aspect of near-field speakers is the fatigue factor of the sound. One reason for fatigue may be the appearance of too many high ranges. The Adam A7 typically lets out more high ranges, which can trick us. On the other hand, it matters that the low range of the speaker is only heard as a hum or a clear frequency. If we are unable to determine the height of the lower frequencies, this may be partly due to design reasons, and the room’s acoustics may increase the problem. A good speaker also emits transients nicely, not in a hollow manner.

Secondary speakers are a must-have accessory in a studio, you may deviate from linearity in the case of these. Other studio types are unsuitable for mixing, these are for post-checks. E.g. Auratone. There are significant quality differences between active and passive speakers. Active speakers usually produce better quality sound. The internal electronics of an active system may seem complicated due to the amplifier inside, but the impulse reaches the speaker via a short cable. This is not the case with passive speakers, the loss is greater and there is no possibility of correcting the transmission, as well as amplifying the low and high sounds separately. For passive speakers, the amplifier can also be an important factor. With an average amplifier, you may not be able to bring out of a speaker what it is capable of. Some of the more expensive monitors are 3-way. This way, the middle frequencies do not burden the subwoofer, making the sound clearer.

Do not choose the smallest speakers, only if you play music as a hobby. If your goals are bigger than that, look from the medium size instead!

Find a linear speaker. You can deviate from linearity to some extent. But it is important that a speaker with disjointed sounds tricks your ears one way or another. Let us pay attention to the material! Speakers with plastic housing are out of the question! Of course, there are many monitors with a plastic front cover, so you do not have to worry about that.

For a secondary listener, look for a less linear speaker or, if the budget is limited, test on a better Hi-Fi device!

The speaker should have a filter switching option between low and high! With some speakers, the low frequencies can only be cut with the filter, not raised! This is not an error, especially large speakers do not have this feature, as these basically contain sufficient bass ranges. If you have a large 8” speaker but your bass is still low, the position of your speaker may be bad.

Larger speakers may also vary because of their size, some of them are slimmer and cannot deliver the vibration perfectly. The value for money in this area is really realistic, so a more expensive, large speaker is likely to perform better.

MONITOR SPECIFICATIONS

Frequency range: (This is the range that the speaker can emit without a significant loss of amplitude.)

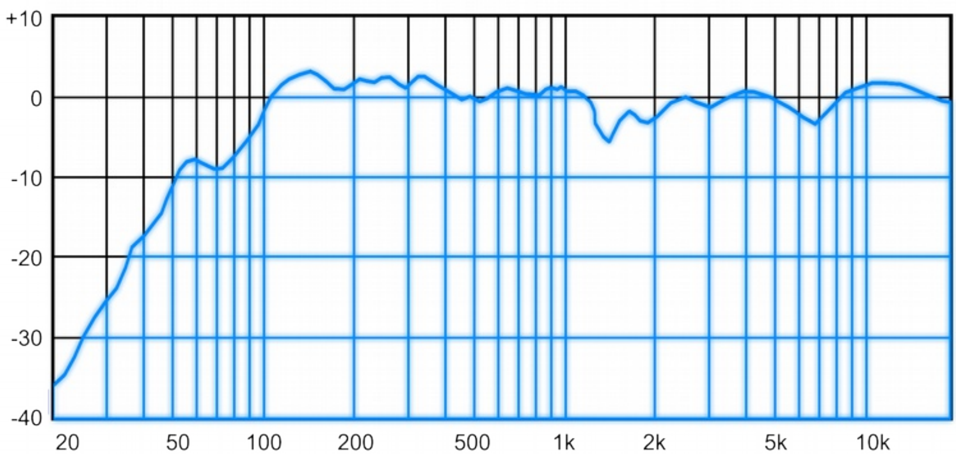
Maximum deviation from linearity: In calculating the value, only frequencies that are stable are calculated. As the amplitude and volume of each speaker at the sub frequencies decreases significantly, I started to calculate this value from the lower frequency transmission value.

Signal-to-noise ratio: The value of the difference between the useful signal and the noise in decibels.

Size of the large diaphragm: The diameter of the low or deep/middle speaker in inches.

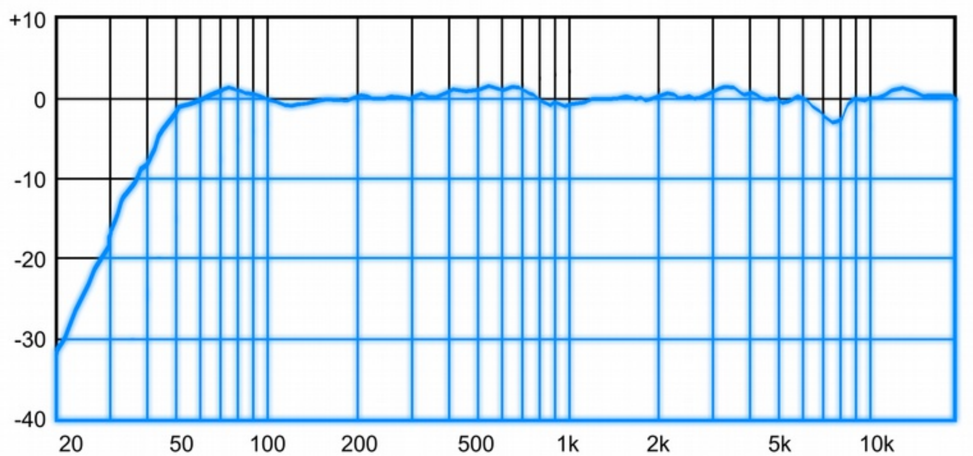
What can we find out from the data? The more linear the monitor, the smoother the frequency curve in the image and the closer the monitor will be to natural sound. The lower value of the frequency transmission, the more the speaker will display of the low frequencies. The larger the speaker's low/mid-frequency driver, the more bass you can emit in the low-frequency range. The smaller the low/mid frequency driver, the better it handles the transients, and the more detailed the signal will be.

M-AUDIO AV 40



Frequency range: 85Hz – 20kHz – Maximum deviation from linearity: 9dB – Signal-to-noise ratio: 90dB – Size of Low/Mid driver: 4”

YAMAHA HS 8



Frequency range: 38Hz – 30kHz – Maximum deviation from linearity: 5dB – Signal-to-noise ratio: No official data. – Size of the large diaphragm: 8”