

Hello! My name is Andrey. I'm a founder of psy trance project SynSUN and electro house project Andi VAX.

I would like to represent you the free video tutorial which is dedicated to secrets of mixing the dance music.

The reasons why I do this were a lot of questions and requests from producers about my way of making and processing the sound.

I saw numerous video tutorials about mixing and mastering music and the main disadvantage is that the creators did an emphasis on ROCK and POP music, didn't pay attention on really important aspects of mixing or explained "how to make" instead of "why we DO exactly THIS".

I didn't find the explanation of some important moments there, so I found the answers on my questions by long-term mistakes and also by discussions with my colleges.

Also in this video were used, in my opinion, the most effective plugins, hardware and software synthesizers for deeper understanding the process.

What I explain here is not a gospel truth. This is only my vision of some aspects in sound recoding. I don't impose on you my own point of view; I just propose it as alternative for achievement of best sounding your tracks.

I want to thank:

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Please sorry for my english!

Introduction

Let's start with basic skills of equalization.

Unfortunately the process of equalization actually is nowhere described and everything is reduced to advices such as "cut dull on 700 HZ" or "add some presence on 1200 HZ". These advices unfortunately are not suitable for real work and sometime leads into temptation, than helps.

So, the equalization...

The equalization allows to eliminate "frequency's" conflicts between instruments. It is connected with nonlinearity humanity hearing, namely with effect of acoustical masking.

The problem of frequency's conflict is that we have two sources with overlapped spectrum and one of them is louder, than another, so we stop to here sound, witch is more silent.

The vivid example of frequency's conflict is the conversation in an airport. When a plane is flying up, we stop to hear what an interlocutor talks. That means that the noise of flying up plane blocks the spectrum speech and we don't hear our interlocutor's voice.

Absolutely the same problem is with instrument's sounds. Every producer has the same problem: all instruments sound perfect separately, but together it is "soup".

This "soup" is a result of acoustical masking.

In acoustic music (for example in symphonic orchestras) the equalization is impossible and a sound of instruments doesn't change.

You can avoid the problem of frequency's conflicts with correct arrangement.

Actually there is much in common between classical and electronic music. For example, Jimmy Hendrix played trance in 70-th years. But that time there were no drum machine and virtual analog synthesizers, so his trance sounded like straight line of bats on acoustic drums and indescribable sounds from his Fender.

In electronic music the equalizers and devices of dynamical processing are using for every instrument in 90 percent of cases.

The main idea of equalization is to eliminate frequency's conflicts.

Main principles

1. There is no universal equalizer!!

Till this moment I haven't found universal equalizer just because of one reason. In music except the process there is a concept of musicality. That's mean that two different equalizers (compressor and chorus) will sound different. Sometime this difference is minimal. But at identical but extreme adjustments of device the difference can be obvious.

In one of my works I was looking for good room reverb. I tried a lot of expensive devices and stopped my choice on simple reverb and from set of standard plug-ins I choose Cubase 4. And the criterion of the choice was MUSICALITY!

Conclusion: learn the way of work and sphere of application of different devices and then define with using the plug-ins (software or hardware)

2. The equalizer works less with cutting frequencies than with raising of it.

3. In the bottom range we use a small width of a strip (Q), in the top range – big width of a strip (Q)

4. Almost any change in one band will affect the sound in the other bands.

5. Acoustical masking is binaural (байнЭрал) phenomenon, theoretically it means, that 2 instruments with overlapped frequencies disturbing each other. Everything would be very sad if no binaural (байнЭрал) unmasking

The sense of **binaural (байнЭрал) unmasking** is if two sound's sources with similar spectrums are situated in one space, than at a parity of levels in 12-15 dB "the weakest" will disappear. But if we dissolve them on 60 and more degrees on a panorama, than "the weakest" will be heard at levels 50-60 dB. So that's mean that it CAN eliminate frequency's conflicts.

Many producers start mixing with pushing a button MONO on a master section. But I use MONO-correction in the end of mixing process for checking correctness of sounding.

More often, the problem with bass and sub-bass spectrum of mix's frequencies can be easily watched in MONO mode. Also a big influence has an addition of right and left channel. If we use working with phase effects incorrect (chorus, phaser, flanger, stereo enhancers, etc), the sound can disappeared from phonogram.

Some video tutorials associate MONO with radio and TV compatibility. I don't exclude this connection, but I think that MONO compatibility lives out one's days, so u have to pay attention on sounding's correction of MONO version mix in within the limits of correct mixing.

6. Easier you can solve the frequency's conflict "on groups" instruments.

It is very difficult to define problematic instruments in a mix without acoustical training.

7. Let's divide a mix into 3 parts. In fact a mix has to be divided into 7 parts, but I do so for you, for better understanding.

So...

1st part: 0 Hz - 120 Hz (in this range are situated only bass and kick)

2nd part: 120 Hz – 8 kHz (in this range are situated all instruments)

3rd part: 8 kHz – 22 kHz (in this range are situated cymbals, hi percussion instruments, high range of all instruments)

According to my words I cut all frequencies lower 100-150 Hz in all instruments and all effects in sends except bass, kick.

For this I use only equalizer uad-1 cambridge eq. It has special mode for extreme cut – Elliptic 6, which promises provides to cut sub bass artifacts for 100 percent.

Also you should cut high frequencies in every synthesizer's part and every effect in sends. I select a frequency from 12 to 16 kHz to my taste.

Equalization in process:

A work of humanity's hearing we can compare with work of spectral analyzer.

Spectral analyzer shows us a total spectrum of all instruments on a certain frequency, but our ear hears only the loudest sound on this frequency.

Our mission is to cut all superfluous frequencies in instruments for each of them sounds in their range.

Conclusion: It is absolutely not important if instruments separately sound awful, important – how they will sound together in mix.

The device spectral analyzer will help you in a process of control (for example DOD RTA, Pinguin, Spectralab or Loud Technologies EAW Smaart).

Spectral analyzer will show on which frequencies spectrums are overlapped (blocked) and also will help which frequencies at what instruments are necessary to cut off. On no account I don't force you to mix with "spectral analyzer", I just recommend you to use your ears.

In difficult situations spectral analyzer helps to control bass and sub-bass. On these frequencies it is possible to reveal problems at wide experience and at big control, I mean special acoustic of a room and qualitative control monitors.

In my home studio I use monitor krk rp-8 and spectral analyzer, which is built in plug-in isotope ozone 3. These means allow to achieve a great result. Finally mastering has to done by specialist in a studio or by label.

In my opinion to do mastering at home is impossible and dangerous. Reasons are the same – you need a wide producer's experience, good acoustic design and great control acoustics, which costs about few thousand dollars for one.

Some my colleges prefer to do the whole process (from beginning till mastering) by their own. With great pleasure I work with session musicians and expert on record, mastering and prefer to give each professional to do their job.

As I said before the easiest way to solve frequency's conflicts is to do it "on groups" instruments. Let's start with "classical group".

BASS + KICK

Kick's spectrum is quite wide striped because of mid-frequency click with expressed "hump" in an area from 40 till 200 Hz.

Bass's spectrum is almost "lineal" in an area from 42 till 1500 Hz. So it's mean that we have a classical example of overlapped (blocked) spectrums.

Let's listen bass and kick together (how it sounds).

The volume of bass and kick must be approximately equally with kick's priority (1-3 dB)

Let's mute of bass channel and remember how sounds kick.

Now we switch on bass again and listen careful what is wrong with kick's sound in low-frequency's spectrum.

Let's switch on/off bass several times to see what happens with kick's timbre.

Usually because of bass's overlapping spectrums the low-frequency's kick disappears.

Solve of conflict #1

A resolution of a problem usually consists of cutting a frequency from a bass for saving "frequency's hole" for low-frequency component of a kick.

ATTENTION!!! Use ONLY non linear phase EQ (for example waves from series Q)

Let's do a small cutting in bass (3-5 dB) and displace from 50 Hz till 200 Hz.

We hear that in one moment a kick begins to sound "wide" in low-frequency's area.

You should cut the main kick's frequency from the bass very careful otherwise the sound of mix on low frequencies would be very faded.

Also you should experiment with right choice of kick's and bass's timbres. At ideal conformity of timbres of a bass and kick, a bass doesn't sound simultaneously with kick, but the impression is made, that it sounds constantly. Often recommend to situate a bass and a kick in different octave. Yes, yes!! You have not misheard!! A kick is same instrument as the others!!! It has their own frequency of sounding, it means, that a kick plays on the certain note. For this reason we cut frequencies from a bass, not from a kick. A kick plays on one note and a bass – on different in the area from one till three octaves.

Also with experience you will understand that a bass's frequency should correspond to a tonality of the whole track. With experience you will learn to hear and to correct the bass's frequency. If you are a beginner usually you hear that THIS kick in THIS mix sounds bad. More often helps a bass sample's replacement on other sample with possible tweaking (in minimal limits).

Also pay attention that duration of a kick should correspond to a speed of track. The kick makes $\frac{1}{4}$ (quarter) from duration of one bar.

The most typical mistake of beginners is a rise of frequency on a bass from 50 till 150 Hz. In a result a kick will sound good but "not in balance" with bass. If we make the kick more silently, we "fail" the rest spectrum of a kick and just aggravate the conflict.

Solve of conflict #2

I'm sure everybody likes when about our track people say "PUMPS"!

Second variant of resolution of overlapped of a spectrum of kick and bass is side chain. Paradox, that from a beginning this method was for quick resolution of overlapping of a spectrum. And now this method became in SPEC EFFECT, which has found very wide application in house music.

The principle of work of side chain is very easy.

We have an operating signal – a kick. And an operated signal - a bass. In insert of a bass's channel we put a compressor. A compressor makes the bass's loudness lower during sounding of a kick. In rock music this effect is used for the resolution the problem of overlapping of a spectrum. In electronic music we get super pumping track! Many beginners want to receive pumping effect, using extreme values of a compression on the master of section. This is a mistake!

Also you should experiment with side chain not only of a bass, but also of other instruments - guitars, leads, pads, percussion loops, etc. Use your imagination!

For finding your own sound in home studio I suggest you to make an experiment with side chain with use of reverb and delay. In fact such simple experiments do your sound UNIQUE!

In my practice I use a package of plug-ins BGTech Dynamics Pack for side chain. Also in new version Steinberg Cubase 4.1 there is a possibility to use some vst3 plug-ins by default in side chain mode.

At creation tracks with other artists I often collide with artificial side chain. This method reduces to simple volume control, so that during kick's sounding the volume of the operated instrument (for example closed hi het) is minimal. This means that in a track appears additional groove and pumping.

NUANCES

In a kick it is usually equalized low-frequency component (kick) and high-frequency (click). The low-frequency kick's "body" can be "dance" - with obviously expressed narrow resonance on 60-150Hz or "rock" with "flat enough" hump from 60 up to 200-300Hz. Usually there is not enough high-frequency component in a kick and so you should add it. This click usually must be added in the area from 5000-8000 Hz. If you add it lower, it can overlap "guitar" frequencies. ATTENTION! You should add the click, when the mix is ready.

With source sound it is very difficult to guess in what range and quantity it is necessary to add a high-frequency component of a kick.

Try to cut off on a live-bass a frequency between 150 Hz and 400 Hz (usually on this frequencies a bass has unnecessary sounds. Often on this frequency we hear a resonance of a room where was made a record of base guitar.

Also for better readability you should add presence on a frequency from 1000 Hz till 1500 Hz. This is better to do in a final mix.

BASS+GUITAR

It is one more group of instruments with an overlapped spectrum and a source of dirt in the lower mid-frequency range of a mix.

As we know a bass's spectrum begins from 42Hz and a guitar's spectrum from 84 Hz.

Do the bass's and guitar's channels by analogy to kick/bass. It can be a "soup" and "indistinctness of sounding" in the bottom middle of a mix. This is it what we were looking for!! Overlapping of spectrums is in the range 150-400 Hz.

The typical resolution of a conflict – cut low guitar's frequencies with shelf, releasing that" a frequency niche" for a bass.

Try to adjust a shelf on an equalizer of the guitar channel on 150 Hz and gradually raise frequency of a cut.

Some time later you will hear that the bass and the guitar "were divided" and we have got ride from a problem of overlapping of spectrums.

After guitar equalization THEY have remained practically "without a bottom". If author's idea - low "evil" guitars, buy thick strings and lower tune of your guitar. It will work much more effectively and more musically, than mixing of timbres of a bass and guitars in a range of 200-400 Hz. A goof guitar track must sound very good from the beginning. It is only possible to cut low frequencies on it.

For sharp sound you can add a frequency in the range from 1200 till 3000 Hz. If a guitar sounds too "acid" you can cut a frequency in this range.

LEADS, PADS & STRINGS

Synthesizers can have different spectrum of sounding. And to have conflicts with every instrument, starts from kick and percussion and finishes with others synthesizer's lots. It is better to check synthesizer's lots "separately" "on compatibility" with kick, bass and guitars. If there is a vocal in a track, you should cut 600 Hz with equalizer. The most frequently mistake of a beginner is that few synthesizers are playing in same octave with similar frequency's timbres. In this situation I recommend you to play your lots in different octaves. You should built up your timbres so that dominating (peak) frequencies were different.

So the conclusion is that the best equalization is no equalization! Better to build up a timbre of synthesizers from a beginning.

Also you can try to cut the frequency's areas, for example 400 Hz in LEAD, 800 in PAD, 1000 Hz in STRINGS.

REMEMBER! It is very important to put correctly in a area ALL instruments with help of equalization, reverb, volume and panning.

I'll talk about it little bit later.

VOICE

Voice is the main instrument in many tracks! It should be always distinct and should block nothing. Many beginners have the same problem with voice's place in a final mix.

Actually it's very easy.

Despite of voice's changing during the singing; it is seldom when a singer's range is more than one and a half octaves. That is why it is possible to consider a voice as the instrument with a static spectrum. The base "frequency's hole" for a voice is on frequency 600Hz (at width of strip EQ in one octave).

This cut is necessary to do WITH ALL instruments, which sounding with voice (bass, guitar, synthesizers) except drums.

For these aims, I create another group track with an equalizer uad-1 cambridge in insert and I direct on it all tracks on which it is necessary to cut out 600 Hz.

Also I create another track with an equalizer uad-1 cambridge in insert and I direct on it all tracks except a bass and kick, on which it is necessary to cut out all frequencies below 125-150 Hz.

But if after this a voice will have problems with some of instruments, do small cut on 1200 Hz – usually it solves all problems.

A voice's equalization is the most dangerous thing. It is very easy to make it "flat", "sharp" and "unnatural". Think twice before equalize it. If you don't like the sounding of the voice, you should change your vocal microphone or your vocalist (hehehehe)! The main idea of the vocal's equalization is cutting by shelf superfluous low frequencies (frequency of a cut is in area 300-400Hz).

Usually there is a wish to make a voice more "brightly" (to add some "brush"), to separate it from other instruments. But you need to do it not with help of equalizer. The classical method is to record a voice to a tape in Dolby mode, but to play it better without Dolby mode.

If you use a digital recording, than you may use spectral processors like Dolby 740 or Digilab Activator.

PERCUSSION AND CYMBALS

The spectrum of percussion and cymbals usually is higher a spectrum of instruments harmonics (from 1000 Hz and higher). But when you use samples of “live” percussion and cymbals a spectrum can be a lot of low- and mid-frequency artifacts which will lead to conflicts to a bass/guitar/synthesizers.

That’s why you should cut with shelf percussion and cymbals low and mid frequencies.

Try to start with 800 Hz and higher.

CAREFUL! Otherwise percussion and cymbals will sound “acid” and “unnatural”.

SNARE

Spectrum of a snare is the largest. It "interrupts" all other instruments, and it is not necessary to make free holes for it.

But you should be careful with an equalization of a snare drum. Classical 14-inch snare has almost lineal spectrum with a "hump" approximately on 200-300 Hz. This is a drum's resonance, which you should remove.

Later it will be easy to "place it in a mix".

In a mix with minimum of instruments it is possible not to do this. Also, with help of default samples, this methodic can be superfluous.

Also almost all well sounding snares it is a combination of a live-drum's sound and a synthetic's sound, or several correct equalizing sounds of live drum kit.

According to nonlinearity of ears the loudest sound from 1 kHz to 4 kHz..

In some cases you should lift this range for subjective loudness of every part (for example, vocal or a small drum).

EQUALIZATION (conclusion).

In conclusion we should make a final mix without frequency's conflict. All instruments should sound in a balance, distinctly "to be read", and it should be no "dirt".

<<<bad vs. good mix comparison example>>>

Compression.

“I want my track to sound loudly!”

The first goal – to mix so that it was loud enough without maximizer on the master bus. It is called “peak’s fight”. To fight with peaks it is possible at the beginning of mix, even before instrument’s arrangement in a room.

At first I decide how I can arrange instruments in space (room) then I cut all superfluous proceeding from quality of sound coloration.

Maximum quantity of peaks in the mix consist of sounds with big contain of high frequently – therefore, instruments of background everything, that over 16-18 kHz u can cut off safely. It is possible not to cut it, if an artistic plan idea of track is obligate to leave high frequencies.

Further. High-frequency peaks on vocal and acoustic guitars. Listen attentive and cut. Here there are more variants already, for example, deesser of guitar is possible too. The main thing not to bring especially heard distortions.

For fight against peaks you can use different limiters and maximizers. What kind of? It’s your choice – it is to each taste and repeated experiments. I think that more musically of all with peaks consults UAD-1 LA2A. I use it on vocal, acoustic guitars and synthesizer subgroups. Some producers prefer to cut peaks with waves L2.

Let’s pay attention that low frequencies make bigger part energy of the mix, but not with such huge quantity of peaks, as high frequencies. So, we don’t touch peaks on low frequencies. Important – to cut carefully everything below 20-30 Hz.

Try to share instruments on subgroups, on frequency attribute and cut peaks of long shot. I’m sure that it will sound more musical, than multiband compressor, but in application all mix.

Our goal: loud and precise mix without any processing on master bus. And NO limiters!

RMS of mix have to be anywhere from minus 10 dB and peaks not loudly than minus 3 dB. It means that a lot of mastering hardware better of all should process this range.

By the way by mixing with master fader on “0” it is like this.

There is an opinion concerning handles of the mixing console. Producers of consoles not on purpose mark position “0” at faders of chanel, because just this position provides the best parity a signal/noise.

We draw a conclusion that if you have possibility to operate the volume of instruments on itself instruments, that better to do it.

Ear can’t catch a distortion presence of peak in size of 44 samples.
It’s one millisecond of sounding at 44100 Hz – Audio CD Standard.

First of all sound sources.

If a timbre dim and incorrectly recorded, then compression can’t help you. It is necessary to write down or reconsider a lot of sounds and to pick up the necessary one.

Sometimes the variant turns out totally unexpected. As a rule, samples and timbres sound absolutely different in the mix and in itself. So I notice once again, you should listen to the WHOLE mix for choose the sound!

Very important is duration of kick, combination with line of bass and etc. But it’s a question of a arrangement, not of a mixing.

Usually maximizes are peak limiters with auto normalize. It suits to final mastering.

It is written a lot of about the theory of a compression.

Everything to know about that –it's a principle of work and what varies at change difference parameters: time of attack, degree of ratio and duration of release. All attributes and effects become visible at the first experiment.

Very interesting are other sort of compression – multiband compression, frequency-selective compression, expansion.

Remember! Good, qualitative compressor works imperceptible. If you use it correctly. If you have to get sound fx, maybe you should use other processing? For example envelope shaper.

A LITTLE BIT about MASTERING.

These tricks carry out special trained stunt masters. Don't try to repeat these tricks in house conditions ;)

In this joke there is a share of a joke, but the rest is the truth.

In usual home studio you can't do mastering faultlessly.

Mastering has to do professional with big experience and good control.

Typical situation when you wrote a track and want produce it....

Serious label always have one or few mastering engineers in own staff. You only need send a mix, Correctly prepared for mastering.

The mane requirement to such a mix: 44.1Khz, 32 bits, WAVE file.

Don't use limiters, maximasers and other devices of processing in the master's section.

If you send several tracks, you must to subscribe and to put the number of each track.

For example:

01 – SynSUN VS ITP – set the pace 32 bit. wav

02 – SynSUN – white and black 32 bit. wav

A little maximizing hint.

Each producer, who respects himself, will never achieve mad RMS his track. It's fate of beginners. At RMS minus 11dB, dynamics of the mix suffer.

But we can use this moment with benefit for ourself! ☺

I discovered for myself wonderful possibility to checkup mix on correct mixing, especially low and sub-low frequencies. The matter is that, if the RMS -8 - -9dB the non correct mixed track to sounds deformed and distorted. But correctly mixed track lose only dynamics. It's a great way to checkup!!!

The most interesting. About fat sound and expensive device.

KICK DRUM

For a long time I use kicks, that I made by myself.
I know three methods of making kick drum.

1st METHOD

Few years ago Erez Eisen (Infected Mushroom) shared with world his secret of the kick drum. You can find this tutorial, if you enter in a web-search “infected kick tutor”. The main idea of this method is the generation of a sine wave with duration 1 second and frequency 50 Hz. Then the sine wave is exposed processing by pitch shift’s effect. Of course I used this method immediately, but unfortunately I didn’t get the result, which I wish, so I didn’t stop! I did the same procedure with preliminary generated sine wave in a sampler Native Instruments Kontakt. I liked this result on all of 100%!

Further, any penetrative kick has a mid-frequency and high-frequency component. Sounds very smart, isn’t it? ☺

Simply: you should add to the kick a closed hi het! A closed hi het you can change to any sample according to your taste. The main task is to add click for better reading in a mix.

Now listen to your favorite tracks – in all of them you will hear the sound mixed to a kick on the highest frequencies.

2nd METHOD

It is a simple work with library of samples.

I urgently recommend you to get Vengeance Sound libraries with kicks, loops, single shoots and fx.

So, this variant of making your hit and recognized kick is a combination of few ready samples. For example you can take the beginning of one kick and the low-frequency release from another. ATTENTION!!! The majority of samples in the Vengeance Sound are specially prepared. If you are fan of crystal-clear sound, make your own samples or find other libraries (my advice is to pay attention on Loopmasters or Ueberschall libraries).

3rd METHOD

Use a synthesizer for synthesize your kick drum.

It can be hardware or software synthesizer or special instrument for the sound’s generation. There were a lot of kick’s experiences. As for me, I didn’t get a decent kick. So I use a compressor and equalizer for it. The best result gave Clavia Nord Modular G2.

The situation is better with special programs.

I recommend you to use: sonic charge microtonic, waldorf attack, ism bassism, creamware kickme.

Using of special programs with “mixing” samples gives a great result!

And remember! If the bass and kick sounds boring, it’s mean that the whole mix will sound same. If the kick and bass are groove and fat, than you should work at the mix and it will be a hit defiantly.

BASS..

It is a very important instrument in the mix. Right sound of bass makes people dancing like crazy on the dance floor and idolize author of the track.

THE SOUND OF THE BASS CHARACTERIZES STYLE OF THE TRACK.

How can we do fat pumping bass?

Don't listen to people who scream that fat bass possibly to produce on expensive synthesizer. Stop! Everything is absolutely not like this. ☺

The main goal - to choose a correct source of a signal and correctly to process it.

As a sound's source is approached almost every analog/virtual analog synthesizer. If you haven't hardware right now, you can use software synthesizer. I recommend Spectrasonics Trilogy, Rob Papen Blue/Predator, Novation V-Station, RGC Audio Z3ta+, Vember Audio Surge, Arturia synthesizers, Logic Audio ES1, Native Instruments Reactor 5, Refx Vanguard.

Remember!!! there is NO ideal synthesizer. There is NO best synthesizer. All of them are good and have their own CHARACTER.

Secret of fat sound of bass hide in right using of parameters of releses/filters sustain and release/amplifier sustain. Often, sustain remove to zero and amaze that you can't hear a bass in intensive mix.

The matter is if sustain is absence, a bright tune you can get only if you control release. The biggest value of release have never practically used. Therefore, the bright tune we hear during several tens milliseconds but it's not enough for a fat bright pumping bass.

Also it is necessary to pay attention to a sequence of a bass - be not afraid to change duration of notes and to use automation.

One more important aspect- it's form of signal. square and sawtooth wave forms are more enriched with harmonics, that's why they sound very bright. You should make all manipulations with sound in the mix, instead of make bass in solo mode.

For processing of bass we use equalizer and compressor. Occasionally it's possible to add a several effects of modulation, but not to the detriment of mono compatibility.

About the equalizer process I told before.

Work of the compressor at bass's processing I should describe in detail.

I read a lot of books and saw many videos, but I didn't understand how to make readable and pumping bass with help of compressor.

Everywhere has led universal receipt: do you want fast attack of bass? That bass punches in chest? Make big time of attack, about 15-35 milliseconds and the time release -40-70 milliseconds.

I have diligently tweak knobs, heard, looked at oscilloscope – the result didn't satisfy me absolutely!

Then I have paid attention on envelope processors. It was the answer on my question.

From existing at present I have tasted practically all similar processors: digital fish phones, creamware spl, samplitude plugins, logic audio enveloper, waves trans-x, but I was impressed only one - Envelope Shaper, standard plug-in from Cubase 4. I use it in send, mixing to the basic signal to taste.

Also as experiment you have to try waves ssl bus compressor. This is compressor gives characteristic click at attack 10 or 30 milliseconds.

Music is creative process, don't forget to experiment!

For example, envelope processor can apply either instrument in the mix.
Try it by yourself!

WIDE STUFF vs OLD SCHOOL

Even more often in modern electro and trance track we hear bass sounding more wide. Modern vision of music in radically differ from experiment of past.

Communicating with many mastering engineers I meet with idioms such as "this bass sounds wader than my control monitors", "this sound is outside of the mix" etc..

Experts became hostages of stereotypes!

I have always spoken for literate disposal instruments in the air.

Establish in your mind room and place there instruments by your taste. IMAGINE IT!

Now I will talk about not simple pan of instruments. I will tell you about effects, which are wide used in electronic music by professionals.

In electronic music there are no rules. Of course you can place kick to left canal and bass- to right (for example as Beatles, Jennifer Lopez) bur there is no reason to repeat other experiments.

By tests, mistakes and experiments were born some methods of unusually wide sound of some timbres.

Listening to SUB6, Ellis Vanghoul и Infected Mushroom, we do not realize ALL work, that they done in ther tracks. All business is in ideal pan, in disposal instruments in the air (in that virtual room) and in the effects for sure.

Now I will tell you how achieve the famous wide sound of Infected Mushroom on synthesizer and unusual effect of the doubled bass of Ellis Vanghoul.

SYNTH DOUBLE TRACK

As I advised you, you should listen to MORE good and different kind of music!! You can gather interesting harmonious and interesting audio special effects. Now I will speak about effects.

When I was listening to Infected Mushroom, I asked myself: "HOW?! How they make such widely sounding tracks without any damage to mono- compatibility.

As EVERY young producer, which writes me, I searched the answers in magic plug-ins stereo enhancers. Of course these plug-ins made some effect of widely sounding, but my intuition said that the secret is easy and it doesn't consist in plug-in.

I choose the right decision – I wrote to author of this music! ☺

You know what is the different between a person without talent and a genius? Except lot's of qualities there is also NO ego and a wish to help other people. Despite of lot's of work, Erez Aizen wrote me quit quick and explained everything in details.

If you know the technology of recording a rock music and you know what does double track means, than the principle is simple.

A guitarist plays a part twice or 4 times. Thanks to unique nuances of playing, same parties have different character and extraction of a sound. Of course on a synthesizer it is difficult and impossible to get different extraction of a sound, but we can get different a character of a sound! I don't mean multilayer sampled libraries; I speak about virtual analog synthesis.

To the left channel we pan an initial timbre, to the right - the same timbre with some changes. What can we change?

1. Filter Cutoff/Filter Resonance in small limits
2. Type of filter
3. LFO's values
4. When you use one synthesizer (for example access virus), change filter routing to SPLIT MODE and change types of filters for the left and right channels. Or direct different oscillators to the different types of filter, panning them till the end to the left and to the right.

What do we receive as a result? - Same timbre, but with different nuances. . Our ear catches a difference of timbres and we hear a very "wide" sound.

Pay attention that for short percussion sounds you can use DIFFERENT samples.

UNREAL WIDE BASS

Now, as promised, we will talk about the widening of the bass.

In the trivial literature about the theory of mixing we can always find information about the positioning of the bass. Only in the MIDDLE!!!. Also, we will find the information about what we CAN'T do with frequencies below 250 Hz. It is not recommended to make experiments with phase lower 250 Hz. But in fact rules are created for them to break!

So.. The majority of producers knows the theory and is familiar with technology of work with plug-ins stereo enhancers.

The user sets frequency and on this frequency makes a manipulation with a phase of a signal. The term phase rotating describes simple shift of one channel concern to another. In practice, we apply a delay of the channel from 10 up to 40 milliseconds. The smallest value of delay gives us effect of modulation as chorus or phaser. The largest value – the delay effect.

We pan an initial sound of a bass completely to the left channel. Also we make a copy of bass, pan it to the right channel and adjust parameter OSCILLATOR PHASE or OSC START. I recommend you to adjust the parameter OSCILLATOR PHASE in MONO mode for saving frequency's mono-compatibility.

So, we get super wide sound of bass. The side-effect of phase manipulations consists in mono-compatibility - the density of a bass in mono is little bit lost and appears effect of modulation in it.

The “wide-bass” tracks are successfully tested in various clubs and on different equipment.

ATTENTION!!! The compatibilities of wide-bass with vinyl's WEREN'T CHECKED!

“Back side of a medal”

There is another side of work with interesting stereo effects – problems with phase. If you have this problem, use Cubase stereo dual panner or waves s1 stereo.

A problem with a phase get to us also from reverbs in sends, effects of modulation (chorus, phaser, etc). In such situations, do what I said before – use stereo dual panner or waves s1 stereo.

DON'T FORGET TO USE PHASE SCOPE!!

PERCUSSION AND CYMBALS

There are almost no rules in panning of percussion and cymbals.

We practically never have deals with real acoustic drumkit and advice as "open hi hat - at the left, ride cymbal - on the right" absolutely not actual.

Nevertheless, for understanding the process I recommend you to familiarize with bases of playing on acoustic drums. Paradox, but some producers pan live drums "on the contrary", means that open hi hat - on the right. In this situation, we listen to acoustic drums from a hall as it is heard by the listeners. Usually live drums pan "from the drummer", I mean open hi hat - at the left.

Don't leave a percussion unprocessed! We don't hear dry sounds in our life. All sounds have small or big echo. Everything depends on a room's acoustic. Try to situate percussion in your imagined room. Use volume, panning, reverberation, work with sound's attack.

Do a simple experience! Clap in your room and listen to echo from your clap. If you clap in airport or subway, you will feel the difference immediately.

Listen and analyze sounds around you.

The real professional work 24 hours per day. I mean not only creation of tracks, but also meditation, the analysis, contemplation ...

For softly cymbals I usually use some chorus. Listen to Vengeance Sound libraries and you will understand what I mean.

VIRTUAL ROOM HINTS

To situate correctly instruments in your imagined room, you should know some simple rules. You should acquire them VERY WELL, once and forever because it is a mortgage of a clear, transparent mix!

To remove the instrument:

1. Reduce volume
2. Cut off high frequencies
3. Cut off more high frequencies, than sounds of the foreground
4. Reflection sound louder than dry sound
5. Big release time for tail of reverb
6. Do not use the enhancer

To approach the instrument:

1. Use enhancer, lifting area high]
2. reverberator must be paned wide
3. reverberator is bright, short and dry
4. Short delay time must be paned wide

The further a source of a sound, the less high frequencies we hear. In situation of mono control, small distances reveals by ears on phase shifts in area of high frequencies. It is necessary to find an equalizer for instrument's "remove" into depth, which moves a phase, simulating analog EQ. And there are no so a lot! We take hi shelf, lower it on 8-10 dB and smoothly lower frequency of a cut. If a sound becomes wide (moves into depth), it means that this EQ approaches to us. I use Waves Q-series. Notice, we don't need linear EQ for our purposes (spatial depth). They are used for other properties. For example they are used for correcting frequency without influence on a spatial arrangement.

Reverberation

Hint #1. In every reverberation there is a Damping. Theoretically, it helps to make a tail of reverberation less artificial, but if plug-ins and instruments are cheap, than it doesn't work. You should cut High frequencies BEFORE they pass thru reverb. For example, create a group and at first place a filter in insert and than – reverb.

Hint #2. Some people think that tails of reflection in a real room are almost similar and it doesn't depend on instrument's position. I'm absolutely sure, that in majority real rooms it is not like this. That means, you should use true stereo reverb.

Hint #3. Many people forget that there are two modes of sends on effects - pre- and post- fader modes. In pre-fader mode the volume of send doesn't depend on fader's place. So, what it gives us? It is very comfortable to make an ambience for many tracks (for example a percussion group – each instruments has own channel) and than to operate the balance of each instrument. The balance changes, but the arrangement - doesn't. Sometimes it is impossible in normal mode to make a reverb tail louder, but in pre-fader you can make it easily.

Hint #4. Each channel has different delay of the processed signal.
For example, left channel has 0 millisecond and right channel – to your taste.
So, we get a wide expansion of a panorama.

Hint #5. There is a good effect: if there is a wide pad or guitars in mix, than good situated on panorama bass and kick become processed.
Try to process wide-striped instrument with a reverberation with algorithm HALL and pay attention that reverberation was “added” to every instrument, includes het and kick.

OUTRO 1

10 advices from ANDI VAX

1. Learn the music's theory! Knowledge of harmony and orchestration do not destroy creative potential
2. Learn to play different kind of instruments for understanding the specific of each of them. For example every boring part of synthesizer sequence can be HIT, because of pitchband, automation, modulation wheel, notes length, and so on.
3. Improve yourself! Start with trying to make your track better than a star's one, imitate. Then improvise, make a track in YOUR OWN style. Try to experiment - this is the only way to succeed.
4. Pay attention on nuances. Ordinary and boring track won't be liked by an audience. If a track is done thoroughly we respect it. Listen to Juno Reactor – it's an example of hard work with the arrangement and the sound.
5. Listen to professionals' opinion. At the beginning, it would be quit insulting. It is impossible to teach a baby to go on foot, at the beginning baby studies to overturn upwards an ass! ☺
6. It is possible to develop a good ear for music. There are special courses of exercise. There is always a hope!
7. There are no rivals between producers. There are colleagues. Realizing this, a great deal can be gained. Not only your colleagues but the whole universe will support.
8. Perform! It's the best way to show your talent, to make some photos for myspace ☺ and get to know some interesting people (musicians, producers & labels).
9. If you have chosen YOUR way – follow it to the end and you are sure to win! Take a look at nowadays world stars. They were background singers!! Other of them was playing for 10 years cover versions of popular songs in garages of their parents and was unpopular.
10. Share your knowledge with other musicians without indemnity. This will speed up the progress and move to our COMMON goal. The most important things in our life get not for money.

Thank you for your wish to learn and experience!

If you have any questions or comments - you are welcome to my web-sites or e-mail me

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