ISSN 1864-6168

THE MUSIC PRODUCTION OF A ROCKABILLY COMPOSITION WITH ADDITION OF THE BIG BAND BRASS SOUND

Logožar R.[1], Lajtman M.[2]
[1]Polytehcnic of Varaždin, J. Križanića 33, HR-42000 Varaždin, Croatia
[2]Lajtman j.t.d., Glavna 2a, HR-40321 Mala Subotica, Croatia

Abstract. *This paper describes the music production of a rockabilly composition with addition of the big band brass sound, and is based on the bachelor thesis of the second author. In the preproduction phase the music genre and style were defined, and the full arrangement was prepared. The standard music production stages, recording, mixing and postproduction, were all done by using the Digital Audio Workstation concept. The recording, including a dozen of the brass sections' lines, is done by the one instrument at a time method. Furthermore, all the lines of a particular brass section were performed by only one player. The richness of the sound could probably be better if more players were involved and if the whole sections were recorded at once, but in that case the afterward time alignment of the individual tracks would be impossible. By taking our approach and by extensive time editing of the brass tracks, an excellent rhythmical accuracy and punchy sound is achieved. The produced song presents a good blending of the rockabilly rhythmical basis, the big band brass sound, and the original ethno-rock style of the band that performed it. The song is about to appear on their new album.*

Key words: *music production, rockabilly, big band brass sound, sound recording, editing, mixing, postproduction, recording of brass instruments one at a time.*

Sažetak. *Ovaj članak opisuje glazbenu produkciju rockabilly skladbe uz dodatak zvuka puhača velikog jazz-orkestra, i temelji se na završnom radu dodiplomskog stručnog studija drugog autora. U fazi predprodukcije definiran je glazbeni žanr i stil, te pripremljen potpuni aranžman. Faze standardne glazbene produkcije, snimanje, miješanje i završna obradba, obavljene su uporabom koncepta digitalne audio radne stanice. Snimanje, uključujući i tuce dionica puhačkih sekcija, obavljeno je metodom instrument po instrument. Nadalje, sve dionice pojedine puhačke sekcije odsvirao je samo jedan puhač. Bogatstvo zvuka je vjerojatno moglo biti bolje da je bilo uključeno više svirača i da se snimala cijela sekcija odjednom, no u tom bi slučaju naknadno vremensko podešavanje pojedinih dionica bilo nemoguće. Uz naš pristup, te uz uporabu sveobuhvatnog vremenskog editiranja, ostvarena je izvrsna ritmička točnost i zgusnut zvuk. Producirana pjesma predstavlja dobar spoj rockabilly ritmičke osnove, zvuka puhača velikog orkestra te izvornog stila izvođačkog sastava, i bit će objavljena na njihovom novom albumu.*

Ključne riječi: *glazbena produkcija, rockabilly zvuk puhača velikog orkestra, snimanje zvuka, miješanje zvuka, završna obrada, snimanje puhača jedan po jedan.*

1. INTRODUCTION

The standard music production generally deals with the recording, editing, mixing and mastering of music. Today it is quite common that the instrumental and vocal arrangements, as well as other creative, very often iterative, phases, like selection of instruments and definition of their particular and global sound, are also intermixed with the recording and mixing process. This is especially so for the production of the contemporary bands without many members, or the songwriter-performer-producer teams who want to have the control of the whole production process. On the other hand, when dealing with more complex compositions, with many people and larger orchestras involved, sticking to the standard music production concept is more efficient. Here one expects that the composition is well defined, that the arrangement is correct and musically functional, and finally, that musicians are prepared to play their parts. Only then we can expect that the standard music production chain will do the rest of the job.

In this article we present a production of a musical piece in which the big band brass sound supports an up-tempo rockabilly composition. The original idea came from the Croatian ethno-rock band **Kom3dija** (named after the Croatian word for *comedy*), who wanted to record a song with added brass sound. In their Megjimurje region such a sound is very popular and played in the folk groups called *bandisti*.[1] The idea was ambitious and brave. Their small budget was supported only by a big enthusiasm and friendship with the local **Čakovec Big Band** — an amateur jazz orchestra composed of talented and educated musicians, many of whom were graduates from music schools.

The idea turned out to be even braver when Kom3dija members decided to do all the recording in the same studio in which they have had recorded their standard-lineup ethno-rock songs. It is a well equipped, but certainly not big, two-room recording facility, called ***Jazbina***, owned by the second author of the paper (Figure 1a and 1b). It has a $5.8 \times 4.7 \times 2.1\mathrm{m}^3$ control room ($Floor - Ceiling\ Area = 27.3\mathrm{m}^2$, $Volume = 57.2\mathrm{m}^3$), and an acoustically treated $4.3 \times 5.5 \times 2.1\mathrm{m}^3$ recording room ($Floor - Ceiling\ Area = 23.7\mathrm{m}^2$, $Volume = 49.7\mathrm{m}^3$). Because of the small height of the recording room, a special attention was paid to providing the highest possible absorption of its ceiling.

[1] Megjimurje region term for the *band members* and the *band*.

Figure 1a. Jazbina recording studio control room.

Figure 1b. Jazbina recording room.

Furthermore, to reduce the number of wind instrument players to the minimum, it was decided that a few best members of the big band would do all the recording on the "one instrument at a time" basis. The procedure common in rock and pop production was now extended to the jazz wind instrument sections.

After all this corner-cutting decisions, a few questions arose. Would the big band brass sound come out rich and groovy? Would it blend into the rock sound and the general ethno-rock concept of the band? Would the final recording suffer because of implementing instruments which are not in the standard lineup and do not participate in, either rehearsals, or in the formation of the song and its sound?

In the further text we address all these questions and explain how they were answered in the course of the preproduction, and the standard music production phases: recording and editing, mixing, and postproduction.

2. PREPRODUCTION

The preproduction phase includes dealing with all the necessary things that need to be done before the actual recording sessions start. In this work we cannot overemphasize the importance of this phase, and, as suggested in the introduction, for the projects of this kind we advocate the standard production procedure.

Before getting into details, a few general things will be stressed. The song's common data are presented in **Table 1**. Among other things, the authors and performers are mentioned here.

Regarding the music production, it was the task of the second author of this article to do all the practical work, starting from the full technical logistics required by the studio facility, through organization and synchronization of the sessions, up to the recording, mixing and postproduction. Above all that, the Kom3dija band assigned him the duty of music producer, a person who would be responsible for the final sound and the recording as a whole. Their trust was based on the cooperation in previous successful projects. They expected that their artistic visions would be fully realized, and if not, the producer was to blame! The first author provided some modest counseling during the mixdown process, and contributed mostly to the academic aspects of the project by mentoring the bachelor thesis [1], and by preparing this paper.

2.1 The importance of arrangement

Thanks to the fact that all the participants of the project were aware of the need for a fully professional approach to the involvement of the wind instruments, the first important steps in this endeavor were accomplished successfully. In this matter, improvisation of any kind would probably be disastrous. Having this in mind, the band first recorded five demo songs to decide which would be the best for adding the brass sound to it. The songs were inspected by the Čakovec Big Band leader and conductor, who was also an experienced jazz arranger (**Table 1**). He discussed his opinion with the producer, and with the Kom3dija band members. Finally, a rockabilly song, which is quite off the band's standard ethno genre, was chosen.

Table 1. The Psycho Billy song essentials.

Song Title:	**Psycho Billy** (In Croatian: Sajkobili*)*
Duration & Tempo:	3min 30s at 220 bpm
Genre:	Rockabilly + the big band brass sound
Words and Music:	**Igor Baksa**
Arrangement:	**Mario Jagec**
Performed by:	**Ethno-Rock Band Kom3dija:** • Igor Baksa – vocal, el. guitars, • Miloš Rok – clarinet, • Neven Kolarić – el. bass, • Marinko Marcijuš: drums, **& Čakovec Big Band members:** • Igor Hrustek – trumpets; • Filip Horvat – trombones; • Mario Jagec – saxophones.
Recorded, edited and mixed by:	Marko Lajtman in *Jazbina* Recording Studio, Lajtman j.t.d. Mala Subotica, Croatia, 2009 – 2010.
Produced by:	Marko Lajtman
Editions:	To be published on Kom3dija album *Prvi svjetski mir* (*The First World Peace*), for Dallas records, Zagreb.
Web availability:	On selected specialized and other sites – after publishing at the end of 2011.

The arranger made *the complete orchestration* for the song, that is, not just for the wind instruments, but also for the electric bass, electric guitars, and the drums. By having the arrangement fully accomplished, we avoided the mistake that is often made by inexperienced musicians and producers who start the recording prior the composition is finished. So be warned: ***the big band sound starts with a properly written music score!*** This is, of course, valid for inclusion of any kind of orchestras, ensembles, or instrument sections containing several instruments, in any kind of music production. Especially so when they appear in music projects as guests. Moreover, in our case the orchestration had to assure a nice and distinctive amalgamate of different musical styles.

The arranger prepared the score, printed the instrumental excerpts, and delivered them to the musicians for practicing. A few rehearsals were conducted by the big band leader before the recording took place. Also, during the recording sessions, he assisted the musicians in finding the right groove and getting the best from their performance. Just before the recording started, it was decided to bring the tempo up from 210 bpm to 220 bpm. This crucial decision helped in achieving the right up tempo feel of the song. The performers had to adapt to the change on the spot. This required slightly higher technical skills and more concentration, but was welcomed by everyone because the groove was improved, and the song finally sat in place.

2.2 The composition's musical foundations

As was already mentioned above, our song is a mix of a rockabilly theme with addition of the big band brass sound. Since the rockabilly is based on the plucked string instruments, primarily guitars, E major tonality came as a good choice --- at least for the rock part of the band. On the other hand, this is not so common tonality for the most of the wind instruments, requiring some adaptation in playing. However, as we shall soon show, a bit strange tonality for the brasses could be creatively used.

The main arrangement parts are composed in the scheme of *questions and answers*. In the main brass theme that starts in the intro, the trumpets play in the four voice harmonies, doubled by the trombones one octave lower. The saxophones answer in a five-voice harmonized theme. During the vocal parts, in their first passage, the saxophones start their short fills. They are backed up by the trumpets and trombones in the second passage (see also 2.4).

In order to avoid masking of the vocal by the brass lines, the effect of *shallow tones* were used in the saxophone parts overlapping with the vocalist. The shallow tones appear when the most of holes on the sax are open, which we refer to as the *hollow positions*. They have less of the basic tone and lower aliquots, and sound thinner than the other neighboring tones. On all saxophones, the shallow notes are A4 – C#5[2]. On Eb alto sax these note values correspond to the concert pitch tones one major sixth lower: C4 – E4. On the baritone sax everything sounds one octave lower than on the alto, so the shallow tones are: C3 – E3. On the tenor Bb sax the notes A4 – C#5 give the tones one octave and one major second lower, i.e. the concert pitch tones: G3 – B3[3].

Good players know how to compensate for the shallow sounding notes, thus making them less obvious. On the other hand, here it can be used to a good effect. Since the song was in E-major, the highest notes played by the alto saxophones could be chosen to be the shallow ones. E.g. in the tonic E-maj chord, the shallow tone E4 appears. In the subdominant A-maj chord the shallow tones are C#4 and E4, and in the dominant B-maj chord it is D#4. In the seventh and ninth chords more shallow tones can appear. If choosing the highest chord tones to be the shallow ones during the vocal lines, the saxes will naturally sound a bit thinner in their lower and mid frequency range. This provides a natural dip in the frequency range important for the vocal presence. On the other hand, when there is no vocal, the fullest saxophone notes F5 – B5 can be played, which on alto sax correspond to the concert pitch tones G#4 – D5. In this way the saxophone arrangement helps in providing of the natural equalization with an important musical function.

Because there is no analogy to the shallow tones on the trumpets and trombones, they were not played on top of the vocal and solo lines. The exceptions were ends of the refrains, where they have the *fp*[4] effect followed by a crescendo.

The hollow positions are also used during the guitar and clarinet solo parts. At first the saxophones play in unison, then in duet, and only at the end of the phrase in the full chords, to avoid interference with the soloist. The trumpets and trombones play only the short accents. At the end of the solo, all the wind instrument lines intertwine with each other to increase the suspense. This is evident mostly at the end of the last clarinet solo, where the short sax accentuations exchange with the trumpets and trombones. This play is furthermore emphasized by panning the two groups of the wind instruments on the opposite sides (see also 4). The end of the song comes in the half tempo, with the wind instruments playing the harmonized boogie-woogie theme and the cadenza in full *ff* (*fortissimo*, compare also 2.4).

2.3 Instruments

The full instrument list is presented in Table 2. The musical basis of the rockabilly composition is played by the rock rhythm section. The drums were played in the rock style. Instead of the upright bass, often found in rockabilly, the electric bass was used to conform to the Kom3dija standard lineup. The electric guitar had the important function of playing a continuous boogie-woogie riff in

---

[2] With middle C = C4, according to the scientific (American) pitch notation. According to the Central European notation, middle C = C1 = C' (= Ger. *Eingestrichenes C*). To convert from the scientific to this notation, number 3 must be subtracted for the octaves with scientific number $n \geq 4$. Below C1 there come: C kleines, C großes, C kontra, etc.

[3] B3 in Anglo-American notation = H kleines in the Central-European notation.

[4] *fp* = *forte-piano*, dynamical specifications of playing forte and then immediately piano. The *fp* is often followed by a crescendo, possibly on the same tone.

ISSN 1864-6168

the classic rock & roll style. It also doubled as a solo instrument close to the end of the song.

The commonly known division of the *wind instruments* is into the two groups:

- *Brass instruments* – trumpets and trombones;
- *Woodwind instruments*, in our case, and normally in the jazz big bands – saxophones and clarinets.

The clarinet is a classic big band woodwind instrument. The saxophone has the same sound producing principle by using the reed as an oscillating element, but is made of brass. Being a kind of a hybrid, it is often omitted from the classical woodwinds (inspect e.g. [2]). In the more precise *Hornbostel–Sachs classification* based on the sound creation and sound formation physical principles, saxophones are the single reed instruments with conical bore [3]. Without going into the details of the nomenclature and genealogy, throughout the text we shall use the standard colloquial terms which are also adopted in the American jazz and pop terminology:

- ***Brass sound***, ***brass section***, or simply ***brass***, to cover all the brassy wind instruments in jazz and other pop bands – trumpets, trombones and saxophones;
- Since the classical woodwinds instruments (clarinets, flutes, oboes, bassoons) rarely play in larger sections in the pop and jazz bands, but mostly appear solo or in duets, we usually speak of them separately.

Another peculiarity regarding the appearance of clarinet in our project deserves a short elaboration. As can be seen from the Table 2, it is a standard instrument in our ethno-rock band, highly contributing to the band's first epithet – *ethno*. The ethno role of the clarinet is to bring the melodic folk tunes with its smooth, noble, and slightly wistful tone, thus making a contrast and counterbalance to the harsher rock sound provided by the guitar-based part of the band. It turns out that, in this song, the clarinet will have two roles: to appear as a big band household instrument, and to present a link to the standard band repertoire and its ethno roots.

Table 2. Instrumental sections and instruments.

Section	*Instrument / Voice*
Rhythm Section	Drums El. bass Hollow body el. guitar, for: – the guitar boogie riffs – the guitar solo
Brass Instruments	Trumpets × 4 lines Trombones × 4 lines
Woodwinds (wood and brass reed-based wind instruments)	Clarinet × 1 line Alt Saxophones × 2 lines Tenor Saxophones × 2 lines Baritone Saxophone × 1 line
Vocals	Lead male vocal

2.4 The song development

The song starts with a classic 12-bar boogie-woogie rhythmical riff played on electric guitar, which establishes the rockabilly sound. The brasses come in with a few chord stabs at the end of the 12-bar intro and, together with a jazzy, swinging, drum break, lead us into the second 12-bar passage. Here, on the top of the boogie-woogie riff, a classic big band swing theme appears with the full brass sound.

After the brass theme resolves with a typical downward glissando and the drums stop on the last two beats, the preparation for the vocal part is made. When the vocal starts, only the guitar riff—accompanied by the bass and drums—continues its rhythmical pattern. This lasts for the whole first 12-bar A chorus. There are no brasses here, so more space is left for the vocal to establish itself and to bring out the melody with the introductory text lines. The brasses join in the second repetition of the A chorus, filling in between the vocal parts. They rise in complexity and power, building up the tension in the third A chorus. At the end of it, the classic 4-bar blues cadence is melodically emphasized and repeated twice into the total of 8 bars — resulting in a part that has the strength and role of a B chorus. Thus the 12 + 4 scheme turned into an 8-bar part that can be viewed as a shortened A chorus, and an 8 bar B chorus. The latter delivers the strongest lyrics line, and repeats it twice[5]. The B chorus brings the climax, the resolution of which concludes the first part of the song.

The song continues with a light variation of the brass swing theme from the intro, and is followed by (only) two A choruses. They bring new, witty and humorous, verses that grab the listener's attention. In the second A chorus the brasses come in and lead us to a classic boogie-woogie bridge. Here, the first 4 bars on the tonic degree are doubled for the total of 8 bars. The rhythm section stops, except for the high-hat cymbal. The instruments hit the *tutti chords*, while the vocal brings the higher tension melody with stronger lyrics. At the beginning of the bridge the chords accentuate the first beats of every two bars, then first beats of every bar, then every two beats within a bar, and finally the acceleration and suspense resolve by moving and continuing the boogie-woogie chord progression to the subdominant degree. The ending of the 12-bar scheme again turns into the climax of the B chorus, which closes the second third of the song.

After having heard the B chorus twice, the song dynamics and brass arrangement go down to mezzo piano, and introduces the clarinet solo. This change of dynamics contributes to the development and dramatic context of the song. The clarinet leads its cheerful Dixieland melody in mezzo-forte, while the brasses grow in arrangement and dynamics. After the clarinet solo, the vocal repeats one (shortened) A chorus which immediately ends in the B chorus. This is followed by two solos, each lasting for the full 12-bar boogie-woogie phrase. The first is a rockabilly-style solo on electric guitar, and the second is a stronger clarinet solo, which now has to fight through the more prominent brasses. In the last passage of the 12-bar scheme the vocal concludes the song by repeating just the B chorus text[5] during the A chorus part. The last four bars are not repeated as before, but are "stalled" by switching to the half tempo, making this last, fourth B chorus, special and more dramatic. The last two bars are a classic boogie-woogie cadenza, being also a standard in Dixie, blues, swing and jazz in general (already mentioned in 2.2). It brings the song to an end, which is fur-

[5] The literal translation of the Croatian line is: "He is my friend, his name is Psycho Billy".

thermore affirmed by a short glissando stab, starting from the tonic E6 chord and going downward in intonation and dynamics.

3. RECORDING AND EDITING

The recording microphones are listed in Table 3. A Tascam DM-3200 digital mixing console served for collecting the microphone signals into its high quality microphone preamplifiers, and for the 44.1kHz/24-bit AD conversion. The computer connection was provided through the IF-FW/DM mkII 32-channel firewire interface. The recording was done in Steinberg Nuendo® 3.2 host program on the PC platform, maintaining the same 24 bit dynamic resolution and 44.1kHz sampling frequency.

The instrument tracks were laid down one by one, starting with drums. As was already mentioned in the introduction, the same recording principle was used also for the wind instruments. But let's start with a short survey of the recording stage according to its chronology.

3.1 The drum kit recording

Both, the rockabilly drums and the big band jazz drums should sound naturally on recordings. So, our goal was to maintain the impression of the natural sounding drums, played by a real drummer, but closer to the rock production. The usual alterations of the drum sound from its direct acoustic appearance—due to the use of close miking, EQs and dynamic processors—are considered as expected by a modern listener. The drums will still be perceived as "acoustic and natural", just more "phonogenic". The processing is applied to make the sound a bit punchier and more up front, and thus more adapted to the recording and reproduction medium.

The Sonor Force 2003 drum kit with metal snare drum was used. The drummer, well acquainted with the song, was governed by the host program metronome at 220 bpm. In the end he halved the tempo as needed. If there had been more subtle tempo changes, we should have them preprogrammed, e.g. by using a tempo track.

Bass drum. The bass drum was recorded with Sure Beta 52a microphone, placed inside the small opening ($\Phi \approx 15$ cm) in the back membrane. The microphone was pushed inside the drum body up to the half of its length, and directed toward the beater hitting zone. Prerecording equalization was done by inserting the Behringer Tube Ultra-Q T1951 equalizer into the signal path, and choosing a peaking curve for +4dB boost at 60 Hz, to emphasize the "thump" part of the bass drum sound.

Snare drum. The industry standard Sure SM57 was put on the snare. It is a classic choice for both PA and studio applications. The microphone was placed at the drum edge, 6 cm above and in parallel to the drum skin, close to the rim (Figure 2). A prerecording equalization was done on the second channel of the tube Ultra-Q, set to +4dB HF shelving boost starting at 3kHz, to enhance the snare component and add freshness to the sound.

Figure 2. The snare drum recording. Shure SM57 is placed close to the edge, in parallel to the drumhead, and 6cm above it. The picture is taken with gratitude from [4].

Table 3. Recording microphones.

Microphone	*Transductor*	*Polar Char.*	*Freq. Range*	*Usage*
 Shure SM57	Dynamic	Cardioid	40 – 15000 Hz	Snare dr., el. guitars, clarinet
 Shure Beta 52A	Dynamic	Super-Cardioid	20 – 10000 Hz	Bass drum
 Sennheiser e604	Dynamic	Cardioid	40 – 18000 Hz	Tom-tom
 Shure Beta 58A	Dynamic	Super-Cardioid	50 – 16000 Hz	Clarinet
 Rode NT5×2 (matched)	Condenser	Cardioid	20 – 20000 Hz	Trumpets, trombones, saxes, overheads
 sE2200a	Condenser	Cardioid	20 – 20000 Hz	Vocal

Tom-toms. There were two tom-toms in the drum kit, the mid and the floor tom, of which only the former was used. It was miked with the Sennheiser e604, slightly to the inside of the rear rim, at the angle of 45^0 to the drumhead axis. The distance from the drum skin was carefully adjusted by listening to the sound and trying to benefit from the microphone proximity effect. No prerecording processing of any kind was done.

Overhead recording. The popular overhead microphone technique is used for collecting the global sound of the drum kit with its cymbals. A pair of well placed microphones, possibly matched, will provide a natural stereo sound image of the drums that is close to how we hear them in a given acoustic space. We applied the recognized *recorderman overhead technique*. It is quite simple and yields excellent results with only 2 or 3 microphones. A detailed description for the recorderman drum setup may be found in [5]. The technique ensures that:

- The snare and the bass drums are well centered in the stereo image, because the distances from each of them to both of the microphones are equal (resulting in the equal loudness and equal phase relationship).

- If the close microphones are used for the snare and bass drum, the interference of their signals with those of the overheads results in no audible problems.
- The captured sound will be close to how the drummer hears it.

One of the important advantages of the recorderman technique is the relatively close position of the overhead microphones comparing to other overhead techniques. This makes it very good for the low ceiling spaces as was ours. There is not much interference from the early reflections and reverberations from the nearby surfaces that are usually introduced when further apart overheads are used. Although our ceilings were made highly absorbent, we still used the above technique for its tight, punchy, and yet natural, sound, that fits our composition's genre.

In our case, a matched pair of Rode NT5 microphones was used for the overheads, bringing the total number of the drum microphones to five.

3.2 The electric bass recording

One of the specifics of the rockabilly sound is energetic staccato playing on the upright bass. As was already discussed in 2.3, here we had a band with a standard rock rhythm section, so that the bass line had to be played on the electric bass. The bassist was supposed to resemble the rockabilly mood by playing the cut-off, or "stopped", notes. He followed the bass line as notated in the arrangement score, and was governed by the rhythm of the recorded drum tracks. The Yamaha RBX 260 electric bass was recorded directly into the Tascam digital console. The equalization was made on the electric bass alone, by adjusting its pickups and onboard passive equalizing filters.

3.3 The electric guitars recording

To get the authentic rockabilly sound, the band leader and guitarist borrowed Washburn J9, a hollow body electric guitar. It appeared on the two el. guitar lines: the rhythmical boogie-woogie riffs, and the solo part.

The following two electric guitar amplifiers were used:

i. Marshal Valvestate 8040 combo, with a tube preamplifier on the so called "boost channel" and a transistorized amplifier — for the rhythmical riffs with just a hint of distortion.
ii. Laney combo full tube amplifier — for a bit more distorted sound on the rockabilly guitar solo.

Figure 3. The placement of the microphone SM57 in front of the Marshall Valvestate 8040 combo amplifier.

The amplifiers' speakers were miked with Sure SM57, as illustrated on Figure 3. As usual, the electric guitar sound was carefully prepared on the amplifiers to be as close to what we need as possible. Additionally, the tube equalizer was inserted in the signal path to give more warmth and to provide in detail prerecording corrections that were not feasible on the amplifiers.

3.4 The wind instruments recording

As was already discussed in the introduction, just one performer presented the whole wind instrument section, and recorded all the necessary tracks ***one by one***.

Trumpets and trombones. These classic brass instruments were recorded with a small diaphragm Rode NT5 condenser microphone. The signal was fed directly into the Tascam DM-3200. The trumpets are the loudest instruments in the big band, quite possibly the loudest instruments at all, capable of producing sound pressures of up to 155 dB_{SPL} at the horn output. This fact had to be taken into consideration, because even the sturdiest microphones cannot accept the sound levels this high. The NT5 has dynamic range of 143 dB_{SPL} at 1 kHz test signal. To accommodate for the trumpet high sound pressures and ensure that no distortion will happen, the microphone was placed at the distance of 80 cm apart from the trumpet horn, 20 cm below the horn axis (Figure 4a).

Figure 4 a and b. Recording of trumpets (a), and saxophones (b). The picture is gratefully taken from [4].

The four lines of trumpets and four of trombones were recorded directly into the host program without any preprocessing.

Saxophones. For all our saxophones the NT5 microphone was used. It's high quality small diaphragm proved to be a good choice for the wind instruments in general. The microphone position was slightly off axis to the right, pointing to the spot in the middle of the instrument body, approximately 50 – 75 cm apart (Figure 4b). Fine adjustments were done for every of the three saxophone types, by carefully listening to their sound.

In total 5 saxophone lines were recorded: 2 alto, 2 tenor, and 1 baritone, as outlined in Table 2.

Clarinet. The typical microphone position for the clarinet recording is shown in Figure 5. The microphone is placed on the side (to the right on the picture), pointing to the lower part of the instrument, but not to the bore. The intention was to get the sound evenly from both, the wooden body, and the tube opening.

Figure 5. Clarinet recording. The microphone diaphragm axis is further off the instrument axis (on the picture to the right [4]).

Although there was only one clarinet line in our arrangement, we recorded it separately for:

i. The parts were it was playing together with the brass instruments — with one Rode NT5, as shown on the Figure 5.
ii. The clarinet solos in the last third of the song — with two Shure Beta SM58 microphones placed in the NOS stereo technique [6]. NOS is similar to the ORTF technique (with distance d = 17cm, and angle $\alpha =$ $= 110^0$) but with greater distance between the microphones and narrower angle (Figure 6). We used the technique to capture more integral clarinet sound.

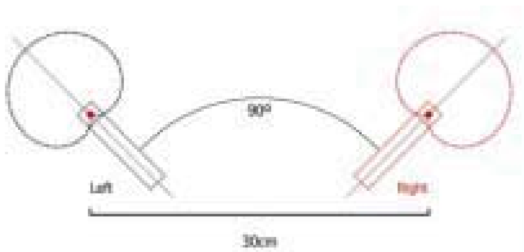

Figure 6. NOS stereo recording technique. The original NOS angle was $\pm 40.5^0$ from the line of symmetry, adding up to 81^0. The practical choice of the angle is 90^0 (the picture is from [6]).

3.5 Vocal recording

The recording of the lead vocal was straightforward. The signal coming from the sE Electronics 2200a large diaphragm condenser microphone was fed directly into the DM-3200 preamplifiers. The singer's voice largely defined the overall sound of Kom3dija band. Here, he had the important task of shaping the vocal interpretation in such a way to artistically unite the band's basic ethno-rock genre with the rockabilly style on one hand, and the jazz style arrangement on the other.

3.6 Editing

One of the key benefits of the one-by-one wind instruments recording was evident in this production stage. Thanks to the capabilities of modern audio recording software, the full and precise time editing could be accomplished. Rhythmical errors on any of the instruments (lines) within the wind instrument sections were carefully corrected. This, of course, would not be possible if the whole sections were recorded.

The editing procedure is very time consuming, but is also straightforward. The nuances and the final effects of the moving of audio parts or audio samples depend on the host program. However, high precision and great musicality are required in the process. The editor must feel what the ideal rhythmical phrase, and the correct "groove" is, and adjust the off-time parts accordingly.

The final results were very good. The groove was smooth-running and swingy, and the achieved rhythmical accuracy was very satisfactory.

4. MIXING

The audio material was mixed in the same host program it was recorded in — Nuendo 3.2. It was done without using any external equipment, according to the well known concept *In the Box* (ITB). It designates that all the tasks are done within a DAW (Digital Audio Workstation), i.e. within the realm of software run on a computer, or possibly the hardware installed in it. In our case it was only the former that took place.

Since the instruments were already recorded with the right sound—as desired or at least very close to the desired—no sound alterations were done prior making the first, *rough mix*. A frequent mistake is to start equalization on individual instrument tracks without hearing the global sound image. This often leads to the counterproductive or excessive use of frequency filters and other processors.

In the preparation of the rough mix, we started with the wind instrument sections by defining the volume relations between particular instrumental lines. Every section was routed to a stereo subgroup for easier further manipulation. Afterwards, the similar actions followed for the rock band rhythm section, and the solo tracks. The result was the definition of the approximate instrument levels and their positions within the stereo image (Figure 7).

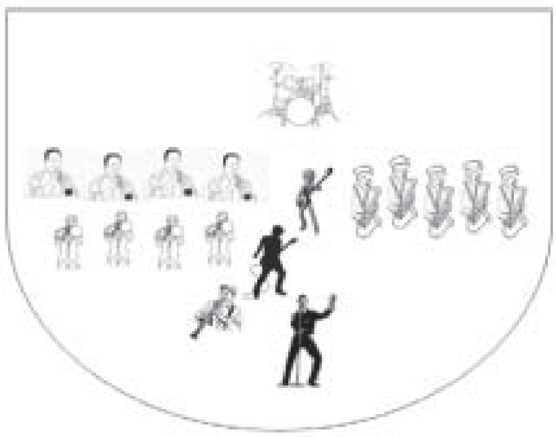

Figure 7. Instrument positions within the stereo image.

After defining the rough mix, the song was carefully analyzed. The defects noticed in the global sound image were attributed to particular instrument sections and their individual tracks. These had to be improved by applying frequency and dynamics modifying tools.

In order to adequately deal with various aspects of the audio material processing, every professional, and particularly the recording, mixing and mastering engineer, should be well acquainted with the frequency ranges of the instrument's fundamental tones and its higher harmonic or overtone contents. One excellent chart containing both of these can be found in [7].

Modifying of the dynamic range is similarly important, requiring knowledge and skills in applying a correct processing to each track. The mixing engineer must be an expert in hearing and analyzing of the audio material, in diagnosing the problems and finding ways to solve them.

All these actions are especially important in the complex productions like ours. Here the good audibility, the definition, and acoustic coordination of all instruments, must be achieved during the recording phase, and even more so in the mixing process.

In both, the frequency and dynamic interventions, we can recognize the following two types of actions:

i. ***Corrective actions***, needed because of the deficiencies in the audio material itself, or because of the way the audio signal will interfere with other signals it the mix;
ii. ***Improving actions***, aimed to make the audio material more *appealing*, where the definition of the *appealing* is, of course, depending on the music genre and style.

Sometimes only the first step will suffice, and sometimes we shall want to make that extra enhancement or produce a special effect. Quite often the same (similar) results can be obtained by taking different routes of action. Which way will be chosen, highly depends on the technical and artistic style, experience, and inspiration of the mixing engineer and producer.

The next step in our production was to set a precise mixdown. Although it contained dozens of separate tracks, we had to organize it in a way that would be easily controllable, and that would not burden the creative process of mixing. Our final mixdown was based on defining the rhythm section and the drums first.

4.1 The drum mixing

The modern drum kit contains drum and cymbal elements that spread a broad frequency range — from the lowest frequencies of the bass drum (≈50 Hz and lower), to the highest cymbal overtones that exceed 16 kHz [7]. Knowing the features and the desired sound of each drum element in a given music genre is essential.

Bass drum. On the bass drum the low and high frequency roll-off is used. The high pass filter (HPF) was set at 27 Hz and the low pass filter (LPF) at 12 kHz, both having the slope of 18 dB/oct. The low range around 60 Hz (precisely at 56 Hz in our case) was slightly boosted to get yet higher thump volume (check also 3.1). To emphasize the kick component of the sound, a high Q peaking boost was applied at 4k Hz. This accentuation of the hi-mid frequencies will help the bass drum to be audible on the (small) speakers with inferior bass response. The Waves transient designer TransX Wide served us to shape the signal envelope (see Table 4). The initial attack was slowed down to make the kick a bit softer.

Snare drum. The snare drum track was treated with the *paragraphic equalizer*, which combines the functionality of the parametric equalizer (adjustable peaking frequency and Q-factor), with the abundance of bands and the appearance of the graphic equalizer (sliding gain potentiometers). Because no physical damping was provided on the snare drum, the resonance of its top skin had to be removed by applying a high Q factor cut by 16.5 dB at 667 Hz. After that the signal is passed through the *saturator* to sound richer. The snare drum sounds great when recorded on the analog tape, as do many other instruments. If the signal is recorded with a high enough level, but of course, not too high, the tape enters its soft *saturation*. This is a nonlinear regime which, like every other nonlinearity, generates distortion. Thanks to the marvel of the analog tape medium, this nonlinear response generates nice harmonics. These days we can find affordable plugins which, at least partially, bring back the warmth of analog recordings to digital studios.

More equalization was done after the saturator. A little boost is made in the lower end at 250Hz. To help the snare get through the mix, its presence is emphasized a bit by boosting the region around 3 kHz. Also, some airy highs were added by applying a shelving equalization at 10 kHz to compensate the loss of high frequencies after the saturator.

The signal was send to the effect track for adding the necessary spaciousness. A software simulation of the Lexicon PCM 96 reverberation unit was used, with the “Chamber” type of space. It is important to fine-tweak the reverberation time and other parameters till the obtained sound blends with other instruments and sits well in the mix.

Tom-tom drum. The tom-tom drum was processed by using the gate dynamic effect to eliminate the unwanted low-level signals picked up from other instruments. The threshold was set at −24.5 dBFS. The gate attack and release times were adjusted to get as smooth, and as unnoticeable dynamic response, as possible. Sometimes, the background signals coming from other drum elements may contribute to the richness of the drum sound and its wider stereo image. However, this is usually not applicable when heavy equalization and post-processing of the drum tracks is applied.

The usage of dynamic gates contribute to the enhanced dynamics, and can, depending on the settings, influence the attack and release times of the recorded sound. In that sense, our choice of using gates is correspondent to the classical way of producing the rock drum tom-toms.

An alternative to the signal gating in nowadays digital production is the manual editing. The unwanted audio signal can be easily silenced out. On analog, and even on digital, tape recorders, this was possible only for longer pauses between the useful signals. Now we can do that in the host recording programs with a surgical precision, regardless of how short the unwanted signals are.

The equalization applied to the tom-tom drum had to reduce its low frequencies, and to stretch its “singing” overtones. The overall guidance was to get a nice, natural sounding tom-tom.

Overhead drum tracks. The two mono tracks from the overhead microphones were routed to a stereo subgroup for the joint processing of both channels. The equalization started by applying HPF set on 500Hz, with the purpose of filtering out the basic frequencies of the bass and snare drum sounds caught by the overheads. The overall drum sound on these two tracks was gently compressed by applying the PSP Oldtimer plugin. It is a simulation of the compressor with fixed temporal parameters (attack and release time), resembling the opto or vintage style tube compressors. A slight make-up gain of 1.5 dB was applied to utilize the extra headroom introduced by the compression.

4.2 The electric bass mixing

The paragraphic equalizer was chosen to eliminate the bass boominess. A low Q ($Q \leq 1$) peaking curve centered at 250Hz was used to make the cut. The harmonic generator was applied to enhance the bass presence in the higher spectrum range, thus helping the bass to get through the mix and to be audible even on the smaller speakers.

4.3 The electric guitar mixing

In the rockabilly style the electric guitars are raw, fresh, and slightly distorted, bringing back the sound of the early days of rock music. The Kom3dija's standard guitar sound was not far from that, only a bit harder, since being influenced by the early punk era. Now the sound was slightly refined and redefined to make it suitable for the rockabilly song (confer also 3.3).

Riff electric guitar. The recorded sound was emphasized by adding the mid and high-mid frequencies. Slight analog tape saturation is added to smoothen out the transients. The low roll-off was applied to filter out the frequencies below 100 Hz. After that the sound was compressed by Softtube FET compressor in its parallel mode. Here the high compressing ratio of 10:1 was mixed with the bypassing uncompressed signal. Short attack and release times were set, and the look-ahead function was turned on to suppress the fast transient peaks of the plucked strings.

Solo electric guitar. To achieve a warm and yet prominent sound of the solo electric guitar, we have applied the analog tape saturation simulator again. Afterwards the sound was brightened a bit by cutting the middle frequencies. Normally the solo guitar needs not a lot in the high audio spectrum. After all, the best guitar amplifier speakers are of limited range, and start their roll-off at 5 – 6 kHz. However, slight corrections, like raising the frequencies above 4 kHz, will help in making the sound brighter and better defined in the mix.

4.4 The wind instrument mixing

The main specific of this work was the implementation of the big band brass sound into the more or less standard rock production. Another thing special was the recording of more than a dozen of wind instrument lines one by one. To achieve the coherence of the tracks on the local and global level, we have always grouped the separate tracks belonging to the same section into one *section subgroup*. Thus we have provided a means for the further ***common mixing treatment of the section as a whole***. Such an approach highly contributed to the natural sound of the brasses in the final mix.

Trumpets. Following the recipe just stated, the separate trumpet tracks were routed to a stereo subgroup, to obtain a homogenous trumpet section. The parallel compressing was used in the limiting regime (compressing ratio higher than 20:1). The attack and release times were set to long values, to ensure the more natural response. Anyway, the wind instruments do not have a fast attack. High frequencies at 8kHz were boosted by 5.5 dB to emphasize the brassiness of the sound.

Trombones. The trombone tracks were compressed individually with a limiter added on output. The individual tracks were then routed to a stereo subgroup. Here the lower mid frequencies from 300 – 400 Hz were cut a little bit, and the highs at 5kHz were raised significantly (+9 dB). Another compressor, with small 2:1 ratio, was put in the stereo signal path to control the sudden peaks.

Saxophones. On the first alto saxophone there was a loud resonating region around G4 concert tone (392 Hz), which is the E5 note on Eb alto sax. It was satisfactory decreased by applying a high-Q cut by 5 dB on the Waves Q10 (the value of 90 corresponds to Q > 10).

Table 4. The plugins used in the mixing process.

Type	*Manufact. & Name*	*Usage*
EQ	Waves Q10 - Paragraphic	• Snare drum, overheads, tom-toms; • El. bass, solo el. guitar; • Saxophones, clarinet; • Vocal.
	Waves API 550A	• Saxophones, trombones.
	Waves VEQ4	• Bass drum
	Softtubes Trident – A – Range	• Snare drum, Tom-tom, • El. riff guitar; • Trumpets.
Compressor	Waves API 2500	• Wind instruments • Postproduction.
	Waves RComp	• Saxees, trombones; • Vocal.
	Waves RVox	• Trombones
	Softube, FET Compressor	• El. riff guitar; • Trumpets; • Vocal.
	PSP Oldtimer	• Overheads
Limiter	Waves L3 Multimaxizer	• Postproduction
Gate	Steinberg Dynamics	• Tom-tom drum
Deesser	Waves RDeesser.	• Vocal
Saturator	Stillwell OligarcDrive	• Snare drum
	SoundToys Decapitator	• El. solo guitar
Transient Designer	Waves TransX Wide	• Bass drum
Harmonic generator	SPL Twin Tube	• Saxophones
	Waves RBass	• El. Bass
Vitalizer	SPL Vitalizer	• Postproduction
Reverb	Lexicon PCM Native	• Snare drum; • Wind instruments; • Vocal.

ISSN 1864-6168

On the second alto sax track the resonance occurred around B4 tone (494 Hz, note G#5 on the alto sax)[6], which was suppressed in the same way as above. To avoid accumulation of the basic tones from the similar instruments, the low frequency range around 100 Hz was decreased by 2 dB. The mid range was boosted at 800 Hz to get the mellower sound. Finally, to obtain more "airy" sound, the frequencies above 5 kHz were raised slightly.

On the tenor and baritone saxes the low-mid frequencies were cut, and the high-mid and high frequencies were raised, with the details depending on the instrument [1]. As before, the individual tracks were routed to astereo subgroup for the common dynamic processing, which helped to homogenize the saxophone section.

The Common Brass Section. The three section subgroups were routed to a *common brass section stereo group*. Dynamic processing, now on the third level, was applied again for better blending of the constituents. The total brass dynamic processing can be summarized as follows:

- *Micro-dynamic processing* applied on the individual instrument level (trombones);
- *Intermediate-dynamic processing* on the instrument-section level (trumpets, trombones, saxophones);
- *Macro-dynamic processing* on the global level, for the brass section as a whole (wind instruments less the clarinet).

The above procedure follows the usual practice of applying several compressing units with more subtle compression in series, rather than just one unit with severe compression settings.

After finalizing the dynamics, the reverberation was added to the common brass section. To keep the brasses close to the listener, a reverberation effect with a short, 1.0 s, RT60 reverberation time, and emphasized early reflections, was chosen on Lexicon PCM Native.

Clarinet – solo parts. Two mono clarinet tracks were routed to a stereo subgroup and equalized by cutting the low range at 125 Hz and 500 Hz. In the reverb applied, the early reflections were emphasized similarly as for the common brass section. However, we selected a bigger space (medium hall), with longer reverberations (1.7s), on the same plugin as above, in order to give more power and spaciousness to the solo instrument.

4.5 The vocal mixing

The Kom3dija lead vocalist is known of his medium raspy voice and an energetic and resolute style of singing. It is equally well suited to various genres, from jazz and rockabilly to punk. During the vocal processing, the attention was paid to preserve the voice vibrancy and dynamics.

The first processor in the signal path was Waves RDeesser set at 5.5 kHz to compress the sibilants. Afterwards, we had to get rid of an unwanted sound component at 246 Hz that somehow entered the recording and was enhanced even further in the later stages of the processing chain. A strong 7.5 dB cut was made on the paragraphic equalizer with a high-Q peaking curve.

The vocal dynamics was governed by two compressors in series. The first one was set to parallel compressing mode, with a high ratio (10:1) compression in parallel to the original sound (compare to 4.3). The second one was set to a mild compression.

Finally, the mono vocal track was routed to a stereo subgroup, with the true stereo Lexicon PCM Native reverberation plugin inserted in its insert points. The Large Wide Chamber space was chosen. The mixture of the input (dry) and the reverberation (wet) signals was to be carefully set in the plugin itself. This is a less standard way of applying reverberations than the usual sending of signal to a common effect track via the send (auxiliary) busses. Namely, in this way the effect cannot be added to any other tracks. However, when the reverb is used for only one stereo pair of signals, as in our case, this is an alternative giving the same final result.

4.6 Overall mixing

After having the complex brass sections already mixed and processed, the final, overall, mix reduced to the complexity of an average rock production. The composition structure and arrangement, presented in 2.4, was to be carefully followed, in order to obtain the desired musical result.

With a detailed automatization we have achieved that the brass sound merges well with the rock band sound. It should be perceived as big and powerful, but must never overshadow the rest of the band. Also, where needed, it had to be adequately subdued to give room for the leading lines, like in the beginning of the vocal parts and during the first clarinet solo. The consistency of the relation between the rock and the big band brass sound had to be carefully preserved throughout the song. The precise arrangement and dynamical playing of the brass sections helped us immensely in achieving that task.

The lead vocal clearly defines the song and reveals the Kom3dija musical style. The voice is upfront and brings the melodic and textual content resolutely, grasping the listeners' attention from the beginning.

During the clarinet solo, the arrangement is quitter, providing a nice relaxation before the grand finale.

In the climax parts during B choruses, in the bridge part, and in the end of the song, the "tutti sound" had to be achieved. However, in difference to the classic big band sound, the electric guitars, the bass and the drums should be also present, reminding us that this is a rockabilly song and that the rock band is still there.

All this demanding dynamics required precisely set automatization. It was programmed in several passages, and in consultations with several participants of the project.

5. POSTPRODUCTION

It is a well known piece of advice that the mastering phase of production should not be done in the same place and by the same people as mixing. This final touch in the complex music production chain should be delegated to specialized postproduction and mastering studios. However, many of the mixing engineers practice to finalize their recordings by providing an ***after-mix*** or ***first-stage postproduction***. Many people consider such postproduc-

[6] B4 = H1, G#5 = G#2, in the Central-European pitch notation.

tion as being a logical closure of the mixing process, and often do it even if knowing that the postproduction and mastering will be applied later again, possibly in specialized facilities. By providing the first-stage postproduction, they, at least temporarily, finalize the audio material and make it better sounding for the radio and other presentations. It is a help to their clients who do not need to go elsewhere to get a final product. Very often it can be a single song, a pilot radio version that, might or might not appear on a future album, etc.

Regardless if the further postproduction and mastering will be done or not, it is a good practice to save the pure mixdown prior any, or most, of the processing is applied. We may be tempted to leave our "super compressor" in the stereo master channel, but only if sure that it is really needed, and that it does not spoil anything. But, in order to be able to repeat the postproduction process later on, either by ourselves or by others, the preprocessed recording should be the starting point. A mastering engineer might require the unaltered mixdown, knowing that her (or his) high quality equipment would do a better job on the material which is not spoiled by previous (inferior) audio processors.

From the above discussion it is obvious that most of the people engaged in audio mixing will find themselves doing at least some stages of postproduction. This is especially so in the small recording studios like ours, where most of the audio engineering tasks must be performed by a single person.

In the after-mix or first-stage postproduction of a stereo recording, we usually assume that the following procedures are carried out:

- Equalization (also including frequency enhancement, excitement, vitalizing, and similar frequency-response shaping processes);
- Compression and limiting;
- Other processing, like sound colorization (by the use of harmonizers and chorus effects), stereo image corrections and expansions, etc.

These actions need not be performed in the above order. Also, they can be performed more than once, or not applied at all. At this postproduction stage we usually deal with only one song or one song at a time. The true album-mastering process must fine-tune the sound images of several different songs, it must level out their mutual loudness, arrange good transitions from one song to another, and assure good overall sound of a music album.

A common rule is that the postproduction should deal only with (subtle) corrections of the global sound image. It is not about correcting the mixdown mistakes, like bad sounding individual tracks or wrong level ratios between them. Also, because we affect the whole recording, only the highest quality audio processors should be used, be they in digital or analog domain. The material should be monitored only on the high quality monitoring systems, and at last, being probably the most important of all, the process should be managed by an experienced person.

In this project we have also provided the first-stage postproduction, which will be depicted here. The final mastering of the album, on which the song is to appear, is planned to be done in a high quality, professional mastering suite. This should prove that we have listened to the advice from the beginning of this section.

For our postproduction, we used Nuendo 3.2 host program. A great help in the procedure is the abundance of good quality mastering plugins available today. By knowing our monitoring system (the active Alesis M1mk2) and our control room acoustics well, and by having gained the experience from many previous postproduction endeavors, we were ready to accomplish this last task.

Fine shaping of the frequency range. The first processor was the SPL Vitalizer, used instead of the standard equalizers. We brightened the global sound in the high-mid region by applying 2 dB boost at 3 kHz, and by turning on the "process" knob of this multifunctional plugin.

Compression. The next processor in the chain was the Waves API 2500 plugin, the software simulation of the renowned industry-standard hardware compressor [8]. It was applied with a low compression rate of 1.5 : 1. The attack time was set to 30 ms in order to preserve the transients and the liveliness of the original sound image. The threshold was set in a way to get just a slight dynamic reduction — not more than 1 dB in dynamically moderate parts, and up to 1.5 dB in the loudest parts of the song. The knee of the dynamic response curve was set to soft.

The DSP implementation of the API 2500 special high-pass filter THRUST®, included in the Waves plugin, was set to its medium option: MED Thrust. The filter is placed before the compressor's RMS detector, which has the all important role of the gain reduction by governing the voltage control amplifiers (VCAs). It provides filtering of the side-chain control signal in order to get a natural, frequency dependable, compression, with response sketched in Figure 8. The ordinate gives the signal that is fed to the RMS detector. With the MED Thrust the HPF starts at the low end and goes up to 200 Hz with the slope $a = +3.01\text{dB/oct} = +10\log_{10}2\ \text{dB/oct} = +10\text{dB/dec}$. Here, oct and dec are *octave* and *decade* frequency intervals, between which the elementary equality holds: $1\ \text{oct} = \log_{10}2\ \text{dec}$. Further on, in the range from 200 Hz up to about 3 kHz, the MED Thrust filter is flat, and up from there the high frequencies are boosted with the same slope HPF as in the bass region. In all, there are three THRUST positions: NORM, flat (working like an ordinary compressor), the mentioned MED Thrust, and LOUD Thrust, with the constantly rising HPF curve with the slope $a = +3.01\text{dB/oct}$ [8].

To explain shortly, the inverse slope from that of THRUST is the well known declining slope of the pink noise energy density: $a_{PN} = -3.01\text{dB/oct}$. This density halves its value after each octave, keeping the total energy per octave constant. The underlying idea of this patented system is to decrease the compression in the lower, and increase it in the higher, frequency region, in accordance to the behavior of the pink noise. This reduces the excessive *gain pumping effect*,[7] which is in many

[7] Compression is usually triggered by the lower frequency signals, which dominate in the standard audio material, just like in the pink noise. The usual compression suppresses all the frequencies equally, so that during compression phase the high frequency signals (including noise) are suppressed for the same amount as the low ones. When the high-level low-frequency

ISSN 1864-6168

music genres, including ours, completely unacceptable. Thus the above mechanism enables louder and smoother recordings which still sound very natural.

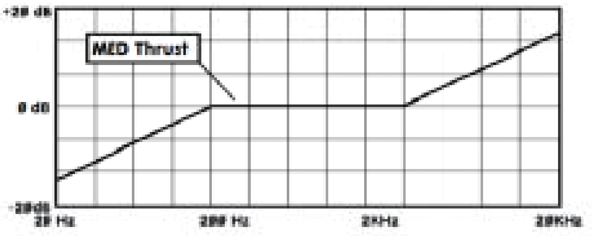

Figure 8. The Waves API 2500 MED Thrust frequency response. From the unit specifications in [8].

Limiting. The last processor in our postproduction chain was the Waves L3 Multimaximizer, a multiband limiter that controls dynamics in five adjacent frequency bands [9]. The limiting threshold was set so that the maximal attenuation did not exceed 2 dB on the loudest passages. The limiters must have a fast response (attack) times to react promptly to sudden signal peaks. In this plugin only the release time was adjustable for each frequency band. The smooth limiting was achieved by rising the release time to maximum, while still maintaining that the attenuation indicators “dance with the rhythm of the music”. The output signal ceiling was set to a value just slightly lower than 0.0 dBFS (-0.3 dBFS), to prevent possible problems that (lower quality) DA converters might experience when reproducing their maximum signal levels.

Together with the previous compression, this mild limiting enabled sufficient dynamical control and the increase of the output RMS level for several decibels, while preserving the natural sound of the final recording.

Dithering. In the final step of our postproduction, when reducing from the 24 bit resolution to 16 bit CD audio format, the now common *dithering process* was applied. It was engaged on our last plugin, the Waves L3.[8]

6. CONCLUSION

Thanks to the development of digital audio technology, the high standards of professional audio production are nowadays more and more achievable in the recording studios that cost only a fraction of what was once regarded as the industry standard minimum. In this work we have described a demanding music production that should prove the above thesis, and also show how small recording studios can be effectively used for the recording of the “big sound” of jazz and similar orchestras.

To accomplish the desired fusion in which an ethno-rock band performs a rockabilly song with the addition of the big band brass sound, it was a must to start from a full orchestration. The band and the music arranger cooperated in the definition of the composition, combining the rockabilly specifics with the power and grandness of the big band brass sound. And while the composer and the arranger had to define the musical ideas, while the musicians had to utter the sound of their instruments and bring in the art of their performance, it was the task of the music producer to amalgamate all this into a professionally sounding product. We have described the process of the music production starting with the preproduction phase, and followed by the recording, editing, mixing and postproduction stages.

The big band wind instruments were recorded one by one. This method, quite uncommon in the music production of the jazz orchestras, was thoroughly and consistently carried out. During the recording, the attention was paid to achieve the correct tonal and rhythmical performance. A precise and comprehensive time editing of the recorded tracks was performed to achieve greater accuracy. The separate brass tracks were summed into sections (trumpets, trombones, saxophones) and processed within stereo groups, in order to make them sound more homogenous and punchier. The sections were then summed into a common stereo brass section, with a rich sound, improved punctuality, and good swinging groove.

Besides this, during the whole song we had to make sure that the “big brass sound” never overshadows the rockabilly foundation of the song. The guitars had to stay in the mix even in the loudest brass passages. On the top of everything, a prominent and stylish vocal had to bring the simple and catchy melody with funny lyrics loudly and clearly, acting like a cohesion factor that united all the composition elements into one.

The ultimate task of music production is to get a frequency-wise and dynamically-wise correct recording, which besides all this must also be musically appealing. We hope that our final recording complies with this requirement, and proves that our production methods could be used as a model for similar musical projects.

7. REFERENCES

1. Lajtman M, *Glazbena produkcija rockabilly skladbe uz dodatak zvuka velikog jazz-orkestra*, završni rad br.68/2009, Veleučilište u Varaždinu, Hrvatska [*Music production of a rockabilly composition with addition of the big band sound*, bachelor thesis, Polytechnic of Varaždin. Croatia].
2. http://www.en.wikipedia.org , articles *Wind Instruments, Brass Instruments, Woodwind Instruments, Saxophone* (last accessed on May 2011).
3. Erich M. von Horbostel , Curt Sachs. 1914. *Systematik der Musikinstrumente: Ein Versuch*, Zeitschrift für Ethnologie, vol. 46, 1914; translated as "Classification of Musical Instruments," by Anthony Baines and Klaus Wachsmann, *Galpin Society Journal* (1961), 14: 3-29.

signal, e.g. caused by a strong kick drum or bass ceases, the compression stops, and the overall gain raises or *pumps up*. The high frequency signals or the higher frequency background noise, being present in the global audio material most of the time, burst out, and causes the “breath-like”, pumping, effect.

[8] Dithering is essential in decreasing the effects of quantization error during dynamic reduction, like when transferring from 24 bit to 16 bit resolution. At low level periodic signals it is manifested in the appearance of artificial periodic patterns, which are heard as added harmonic content, i.e. the digital harmonic distortion. To prevent this, a low-level noise is summed to the original signal to “disturb and spoil” the harmonic patterns. The result is a less audible distortion, since the ear is less perceptive to the low level noise than to the same level harmonic content.

4. David H, Robert R, *Modern recording techniques*, Focal press, 2005.
5. http://www.hometracked.com, inquiry: *Recorderman overhead drum mic technique* (last accessed on June 2011).
6. http://en.wikipedia.org, article: *NOS Stereo Technique* (last accessed on June 2011).
7. http://www.independentrecording.net/irn/, *Interactive Frequency Chart* (last accessed on June 2011).
8. http://www.apiaudio.com/2500.html, *API 2500 stereo compressor* (last accessed on June 2011).
9. http://www.waves.com/Content.aspx?id=255, Waves L3 Multimaximizer (last accessed on June 2011).

ACKNOWLEDGMENTS

The authors wish to thank all the participants of this project: the Kom3dija members and their leader Igor Baksa, the wind instrument players from the Čakovec Big Band and their leader Mario Jagec.

Logožar Robert
Polytehcnic of Varaždin
J. Križanića 33, HR-42000 Varaždin, Croatia
robert.logozar@vz.t-com.hr

Lajtman Marko
Lajtman j.t.d.
Glavna 2a, HR-40321 Mala Subotica, Croatia
marko.lajtman@ck.t-com.hr