Technics Digital Amplifier
Technical Description

Content

1. Intro

Conventionally, analog systems have been the mainstream for high-end audio amplifiers, and the technology for achieving high-quality sound has a long history, and the development of circuit methods and high-quality sound components has been almost completed. Especially, in the circuit system, the negative feedback technology (NFB) is widely used to reduce the distortion due to the non-linearity of the linear region of the transistor and the crossover distortion of the push-pull operation.

Although it can be suppressed, since the audio signal itself is also fed back at the same time, an audio signal with a time delay is subtracted from the input audio signal, which causes a problem of degrading the transient characteristics of the music signal. Some dislike it, and non-feedback type amplifiers or amplifiers that detect and feedback only distortion components have been put to practical use, but the output power cannot be big, and the accuracy limit of distortion detection and complicated circuits also have problems. Originally, beside the low power efficiency, the analog method has the drawback that

the quality of the sound is likely to deteriorate distortion caused by component inaccuracy, and noise mixing due to the complicated circuit, wiring, structure, etc.

On the other hand, thanks to the evolution of digital technology and devices, it has become possible to realize a digital amplifier capable of high-accuracy signal processing and transmission that cannot be realized by an analog amplifier. Looking at the operation of the output transistor, there is also the merit that a high-quality amplifier can be realized without relying on the NFB, by performing switching operation instead of using a linear region with a lot of distortion like an analog amplifier. In terms of switching, the energy generated by the power supply as it is and supplying it to the speaker, there is an advantage that the speaker can be driven more efficiently than by an analog amplifier that converts it into heat inside a transistor.

In the sound source, digital data such as CDs and net audios had become the major. And for such music sources,

it is considered appropriate to be able to faithfully transmit the digital sound source in digital form right up to the speakers. Furthermore, with the higher performance of peripheral devices such as high-speed A/D converters, even if the analog sound source is once converted to digital, there is almost no deterioration in sound quality, and the merit of the digital system is becoming more advantageous. We believe that digital amplifiers will be the mainstream amplifier technology in the future – even with high-grade applications.

2. Full Digital Amplifier

Unlike analog, digital signal processing should be free from the generation of noise or distortion, but the only drawback is jitter. Jitter is the fluctuation in the time interval of digitally transferred data, and the output signal is distorted due to the non-uniformity of the time interval of the clock that controls the timing of digital data transfer, resulting in deterioration of sound quality.

In conventional digital amplifiers, the measures against this jitter have been insufficient, so that the bass sound is not present or the sound is "tuned" rather towards the trebles, and it has been shunned as a so-called "digital smell sound". Technics has realized an amplifier that can take advantage of digital signal processing by overcoming the drawbacks of conventional digital amplifiers by providing sufficient treatment for this jitter.

The digital amplifier is a method of driving the speaker by turning ON/OFF the output transistor with a high-speed PWM signal, and the accuracy of the output signal is determined by the accuracy of the PWM signal. Generally, the precision of the PWM signal can be secured by using a clock generated with high precision by crystal oscillation.

But the digital audio signal input is e.g. read from a disc or obtained from the network in the source devices, and transmitted digitally by optical or coaxial cable to the amplifier. Therefore they are accompanied by jitter, and they are asynchronous with the clock used in the amplifier. In order to realize a high-quality digital amplifier, it is essential to be able to generate an accurate PWM signal from these asynchronous and jittery inputs.

There is a commonly known method called Class-D amplifier, and Technics digital amplifier is often mistaken as Class-D, but strictly speaking, Technics digital amplifier is not Class-D. Class-D amplifiers obtain a PWM signal by comparing a sawtooth wave with the analog input signal by a comparator as shown in Fig.1, and drive an output transistor with it. It seems that the output stage is a switching amplifier configuration, and "D" is mistaken for a digital acronym and is recognized as "digital amplifier = Class-D", but D of Class-D is Class-A, Class-B, Class-D is used as the one following to Class-C, not digital amplifier = Class-D.

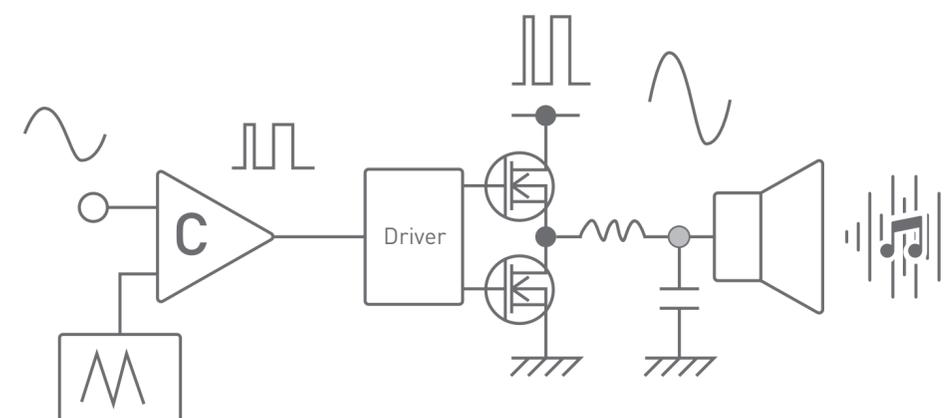Class-D amplifier is an amplifier method that applies

Fig.1

analog signal technology such as sawtooth wave generation and comparator, so it has the drawbacks of conventional analog amplifiers such as the possibility of distortion and noise mixing depending on the accuracy of parts etc. Further, since the input is an analog sound signal, it is necessary to once perform D/A conversion on the digital signal, and there is a limit to accuracy in handling a high precision signal such as High-res sound.

Technics' full digital amplifier processes digital input as it is with high precision, so it is advantageous especially when handling high-res signals. When dealing with analog signals, a 192 KHz / 24 bit high-precision A/D converter is used to ensure high performance.

3. Block Diagram
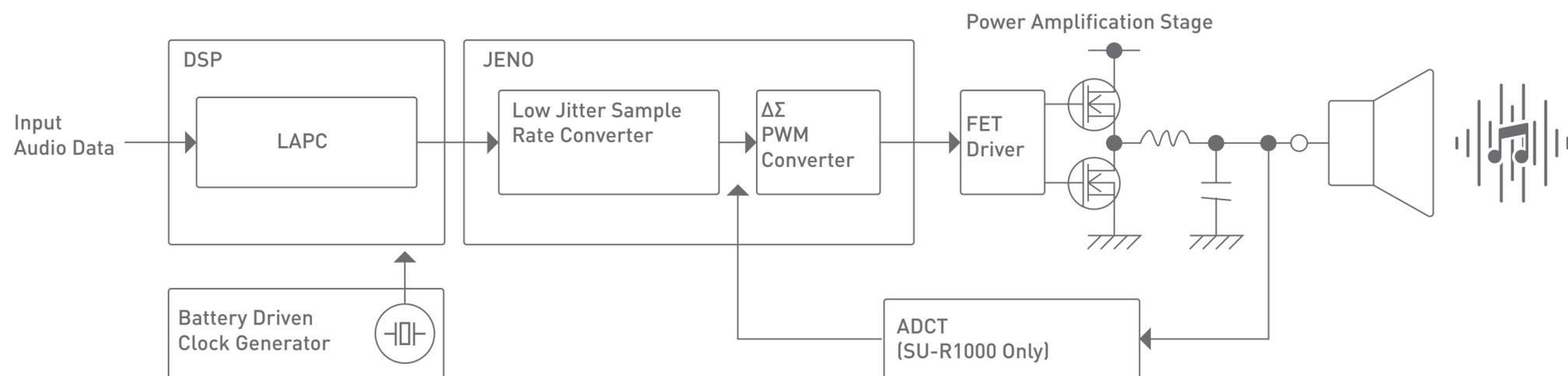
Power Amplification Stage

Fig.2

Fig.2 shows the configuration of a Technics digital amplifier. In the figure, the Input Audio Data is a digital audio signal that is first sent to the DSP to get LAPC processing and then input to the JENO. The JENO is the core block of a Technics amplifier, which consists of a Low Jitter Sample Rate Converter and a ΔΣ PWM converter. It converts the input audio signal into a high-precision, high-quality PWM signal and feeds it to the output power amplification stage.

The output stage is a full-bridge switching amplifier, and the 768KHz PWM signal is turned ON/OFF by the output transistor. (In the explanation, it is described as a half bridge for the sake of simplification of the figure.) Especially in the high class machine (SU-R1, SU-G30, SU-R1000), the output transistor here has an excellent high-speed switching performance with low impedance. GaN (Gallium Nitride) is used to achieve highest sound quality.

LAPC (Load Adaptive Phase Calibration) is also a function unique to Technics, and it adjusts the gain and phase frequency characteristic of the output signal potentially influenced by the speaker impedance, and realizes flat gain and phase characteristics at the speaker terminal, regardless of which speakers the amplifier is connected to. The precision of the PWM waveform highly affects performance of the digital amplifier, and the precision of the clock greatly contributes to this. The clock is usually generated by crystal oscillation, and the accuracy of the oscillation is greatly affected by the quality of the power supply used in the oscillation circuit. Therefore, Technics eliminates the noise generated in the rectifier circuit by driving the crystal oscillation circuit with a battery, and has realized more accurate crystal oscillation.

Furthermore, in the SU-R1000, ADCT (Active Distortion Cancelling Technology) is newly developed to eliminate the distortion generated at the power amplification stage, thus higher sound quality is realized. Each of these high-quality sound technologies will be described below, but the ADCT will be described in detail in a separate document along with other technologies equipped in the SU-R1000.
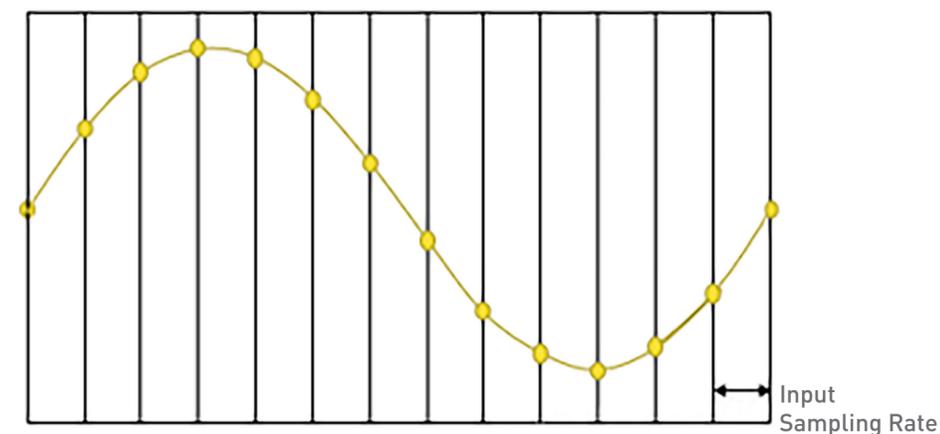
4. Jitter

Fig.3-1 Original Data

Fig.3-2 Distorted by jitter

Fig.3-3 Up-sampled data

Before the description of the detailed operation of each circuit block, the problematic jitter will be explained with reference to the drawings. Fig.3 shows digital audio signal. In each figure, the horizontal axis is the time axis and the vertical axis is the signal value. Fig.3-1 is the original digital audio signal, and the sine wave is sampled at "Input Sampling Rate", and the values sampled at each timing are shown by yellow dots. The signal sampled in this way is recorded on a CD or stored on a server, and that signal is read out, converted to a sampling frequency of high frequency (768 KHz), converted to PWM, and output.

This is the operation of the digital amplifier, but the time axis of the digital data actually input to the amplifier fluctuates depending on the disc reading and transmission path conditions. The fluctuation is extremely illustrated in Fig.3-2, and the Input Audio Data in Fig.2 is such a signal. Since the signal, originally supposed to have the timing indicated by the vertical dashed line, is displaced to the vertical solid line position, the audio signal is also distorted as indicated by the red line. It should be noted that, for the sake of visibility in the explanation, there are extreme time shifts in the figure, such a large time shift does not occur in the actual signal, but the influence on the sound quality is large.

The signal shown in Fig.3-3 is the signal with the sampling frequency converted to a higher frequency (768 KHz) while removing the distortion due to this jitter, which is the output of the "Low Jitter Sample Rate Converter" in Fig.2. Then, PWM conversion is performed to drive the output transistor.

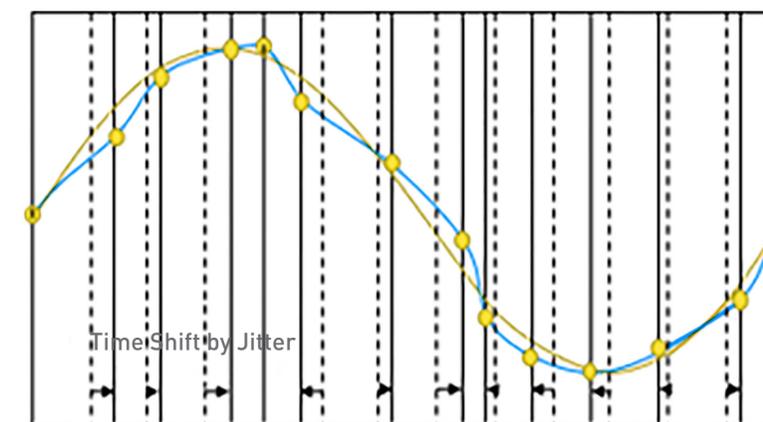# 5. JENO (Jitter Elimination, Noise-Shaper Optimization)

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┌─ JENO ─────────────────────────────────────┐
│                                             │
│   ┌──────────────┐      ┌──────────────┐    │
│   │ Low Jitter   │      │ ΔΣ           │    │
──────▶ Sample      ──────▶ PWM          ──────────▶
│   │ Rate Converter│     │ Converter    │    │
│   └──────────────┘      └──────────────┘    │
│                                             │
└─────────────────────────────────────────────┘
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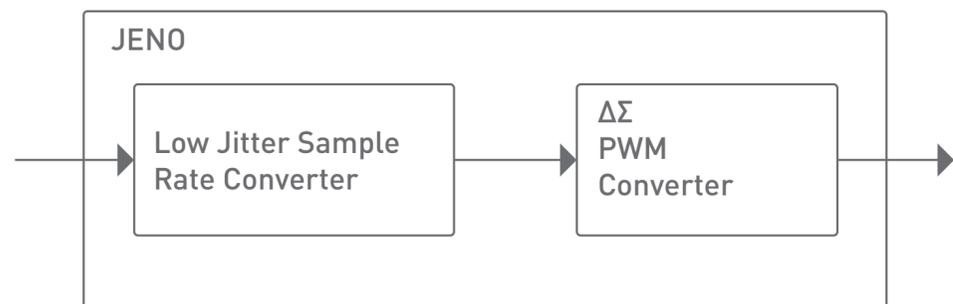
JENO is the core block of a Technics amplifier, which takes various types of digital audio signals, removes jitter, controls volume, and then converts to high-quality and high-precision PWM signals.

Depending on the model, all circuits are implemented in a custom LSI (Fig.5) or FPGA* (Fig.6) at high density, shortening the signal path inside the set and reducing the operating noise of digital circuits. The following describes each block.

*Field Programmable Gate Array

JENO Engine搭載LSI

Fig.5

Fig.6

5-1. Low Jitter Sample Rate Converter

The digital audio signal input to the "JENO" is first input to the "Low Jitter Sample Rate Converter", and the sampling frequency of the sound source input at various sampling frequencies is converted to the single PWM drive frequency (768 KHz). In order to perform PWM drive with high accuracy, the signal must have no time fluctuation. Therefore, the PWM signal is generated from the highly accurate clock signal obtained by the crystal oscillator.

On the other hand, the input signal has various sampling frequencies such as 36 KHz / 44.1 KHz / 48 KHz / 96 KHz / 192 KHz, contains jitter due to the influence of the transfer path, etc., and is asynchronous with the output. Therefore, the Sample Rate Converter is required to have a function of converting the sampling frequency, that is oversampling, while removing the input jitter.

Fig.7 shows a block diagram of the "Low Jitter Sample Rate Converter". In the figure, the "High Precision Over Sampler" outputs a 786 KHz digital audio signal by highly accurately over-sampling the input digital audio signal

having various sampling frequencies.

Generally, an oversampling filter performs up sampling of input data to a sampling frequency that is the least common multiple of the input/output sampling frequency, and then decimate the data at the output sampling rate. Since this oversampling signal processing accuracy determines the performance and sound quality of the entire amplifier, it is necessary to ensure sufficient accuracy.

However, in this case, the sampling frequency of the input and the output may not be a simple integer ratio, and since the input also includes jitter, simple decimation cannot be used. Therefore, the "High Precision Over Sampler" realizes high-precision filter performance with a stopband attenuation of -180 dB with a multi-stage filter equivalent to 108256 taps, and at 768 KHz PWM frequency by 1024 times oversampling and neighbourhood value interpolation. Thanks to this processing, sufficient time accuracy of less than 1ppm is secured.
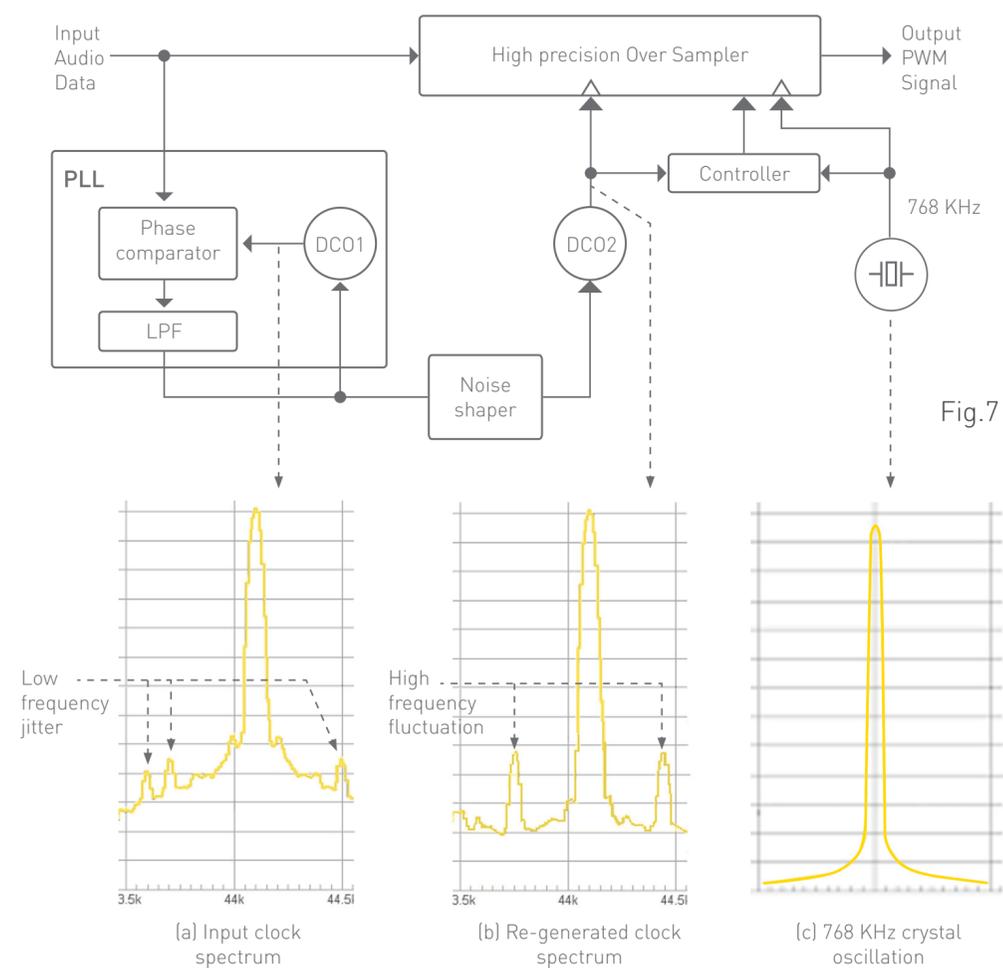

Fig.7

(a) Input clock spectrum

(b) Re-generated clock spectrum

(c) 768 KHz crystal oscillation

Fig.8

Equally important as the oversampling accuracy in the "Sample Rate Converter" is the accuracy of detecting the sampling frequency of the input audio signal. If this accuracy is low, correct oversampling operation cannot be obtained. Therefore, Technics has developed a unique method that applies "Noise Shaping" technology to the conventional digital PLL.

In Fig.7, the "Controller" controls the operation of the "High Precision Over Sampler". It calculates the oversampling condition from the relationship between the input signal and the output clock (768 KHz), controls the "High Precision Over Sampler", and controls the correct output timing.

Actually, the oversampling ratio is determined by detecting how many input samples are input within a fixed time of the 768 KHz clock generated by the crystal oscillation. Generally, the "PLL" in Fig.7 extracts the clock ("DCO1") from the input audio signal and uses it to detect the sampling frequency. Here, the "DCO1" is a digital variable frequency oscillator, and the output of DCO1 and the Input Audio Data are phase-compared, and the component through the "LPF" (loop filter) is the clock information, which is used to control the oscillation frequency of the "DCO1".

However, as shown in Fig.8(a) "Input clock spectrum", the time axis here fluctuates widely due to the influence of the transmission path, especially in the low frequency. If a frequency is detected using this, a fluctuating sampling frequency appears in the vicinity of the true sampling frequency with a high probability, and it is difficult to accurately obtain the correct sampling frequency. Therefore, a "Noise Shaper" is applied to the clock information ("DCO1" control signal) to convert the low frequency noise component (that is, the frequency fluctuation component) to the high frequency range.

The "Noise Shaper" is a technology realized by the $\Delta\Sigma$ converter that pushes the quantization noise into the high frequency range to reduce the low frequency noise. In the audio technology field, it is applied to D/A converters and PWM/PDM conversion. Here, the technique is applied to the control of a DCO (Digital Variable Frequency Oscillator). A "Noise Shaper" is applied to the control signal, and another digital oscillator "DCO2" is driven, as shown in Fig.8(b) "Re-generated clock spectrum". In addition, it is possible to obtain a clock with low-frequency jitter reduced to a level equivalent to the crystal oscillation accuracy. When this signal is used, the probability of occurrence of values near the true sampling frequency is reduced, so that the sampling frequency can be obtained more accurately, and the "High Precision Over Sampler" can be controlled correctly. The jitter of the input signal that has moved to a higher range can be absorbed by the multistage filter in the "High Precision Over Sampler", so it is possible to realize a method with extremely high precision frequency detection and jitter suppression.

In this way, a highly accurate Sample Rate Converter can be realized by the unique method of adapting the "Noise Shaper" technology to the clock information of the "PLL", which makes it possible to improve the sound quality of the digital amplifier.

5-2. ΔΣ Noise Shaping Optimization

The output of the "Low Jitter Sample Rate Converter" is input to the "ΔΣ PWM converter" block via the volume control circuit. With analog amplifiers, volume processing is performed at a relatively early stage of signal processing, but with Technics digital amplifiers, the input audio signal is processed as full bits up to that point in order to ensure the maximum conversion accuracy of the "Low Jitter Sample Rate Converter". However, volume processing is performed immediately before all processing is completed and conversion to PWM is performed.

In the volume control block, a 32 bit 768 KHz audio signal is multiplied by a volume coefficient to obtain a 32 bit (40 bit in SU-G1000) output. The output is sent to the "ΔΣ converter" and becomes a 7-bit intermediate signal noise-shaped at 1.5 MHz (SU-C700 keeps 768 KHz), which is converted to a PWM signal by the PWM converter.

The noise shaping here ensures a S/N ratio of -90 dB for the C700 and -120 dB for other models, as shown in Fig.7.

By increasing the multiple rate of oversampling, it is possible to realize an amplifier with a wider band and a lower noise floor (for example, up to 3 MHz), but the PWM frequency also rises and the operating speed of the output transistor described later becomes a problem. It employs 1.5 MHz operation, which is the limit of usable output transistors.
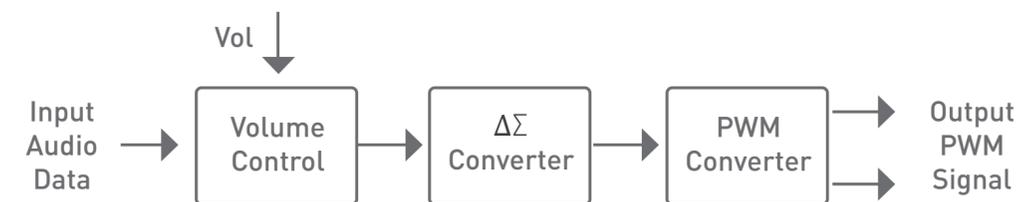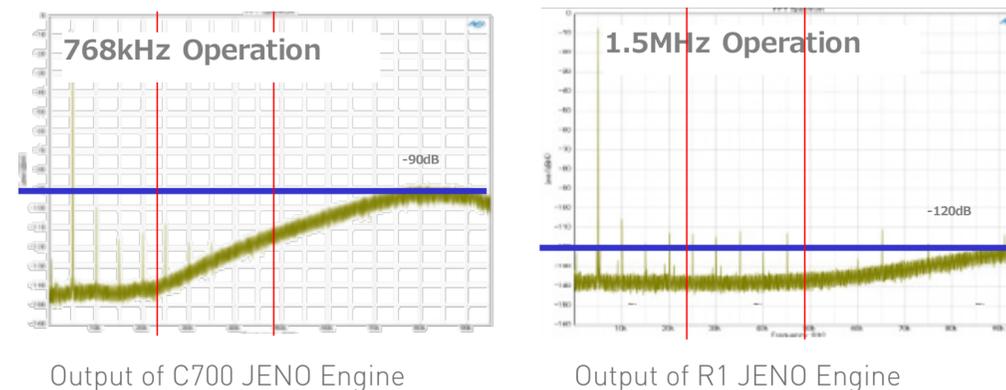

Fig. 9

768kHz Operation — -90dB

1.5MHz Operation — -120dB

Output of C700 JENO Engine

Output of R1 JENO Engine

Fig. 10

Fig.11

The noise shaper is applied by the "ΔΣ converter" and the audio signal up-sampled to 1.5 MHz is sent to the "PWM converter". Since the output circuit of the actual Technics amplifier is a full bridge configuration, the + side and-side of the speaker are driven by separate drive transistors as shown in Fig.11.

As shown in Fig.12, the "PWM converter" generates a complementary ternary signal (2 bit) PWM signal from the 1.5 MHz "ΔΣ converter" output, and each drives a +/- driver. With this, the frequency of the signal that drives the output transistor is lowered to 768 KHz, and the speaker drive operation of 1.5 MHz is realized while reducing the influence of the response delay time of the transistor.
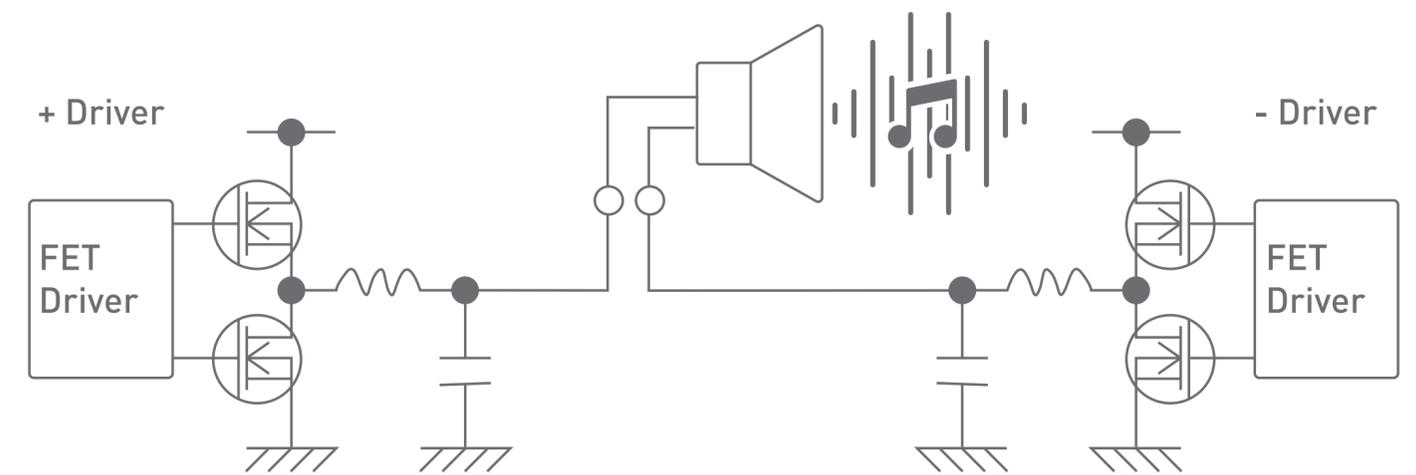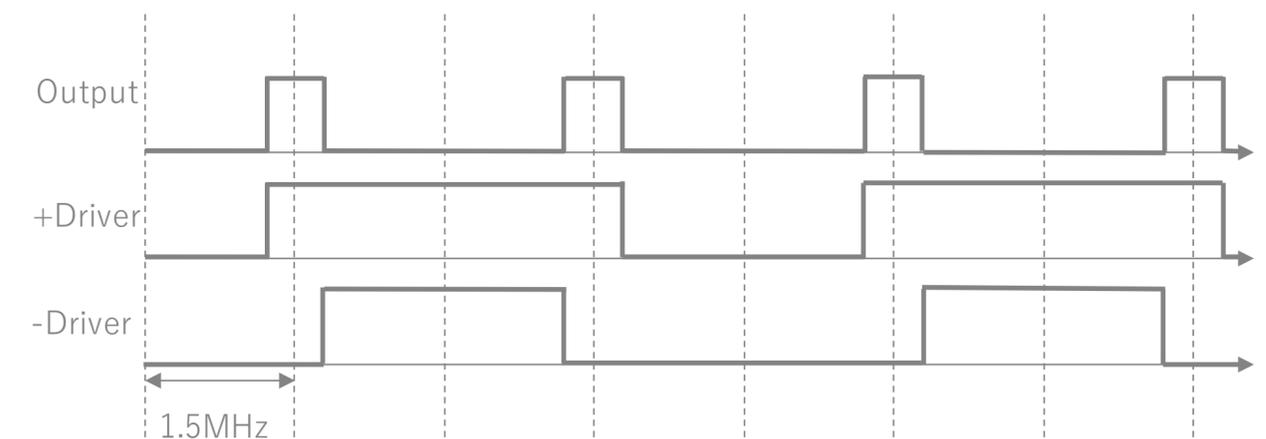
Fig.12

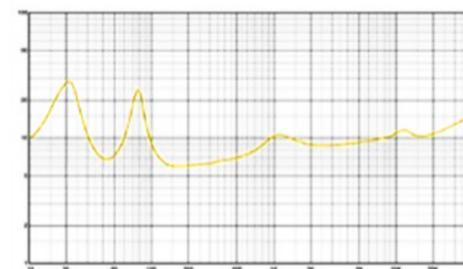# 6. LAPC (Load Adaptive Phase Calibration)

SPEAKER IMPEDANCE (EXAMPLE)

Fig.13

Fig.14

LAPC MEASUREMENT PROCESS

Fig.15-1

PLAYBACK PROCESS

Fig.15-2

The impedance of the speaker is not constant as shown in Fig.13, but has frequency characteristics, but ideally the amplifier needs to drive the speaker without being affected by the characteristics of this impedance. Especially in a digital amplifier, as shown in Fig.14, an LPF (low-pass filter) for the removal of the carrier signal is required after the output transistor, so the frequency characteristics are disturbed more than in an analog amplifier due to its effect. Of course, these can be improved by applying negative feedback (NFB) technology, but when NFB is applied, there is a side effect that a signal including a sound signal is returned from the output to the input, which causes distortion in the transient, and the sound quality

of the digital amplifier may be damaged. In order to solve this problem, Technics compensates the characteristics by measuring the output gain and phase characteristics of the amplifier at the speaker terminal with the speaker connected, and ideal impulse response secured for any speaker. We have developed an algorithm that realizes such processing, and that is "LAPC".

Fig15-1 shows the characteristics measurement and playback operation of "LAPC". First, the user initiates the "LAPC Measurement Process" on the amplifier by pressing and holding the dedicated button on the remote control, or the button on the front panel depending on the model.
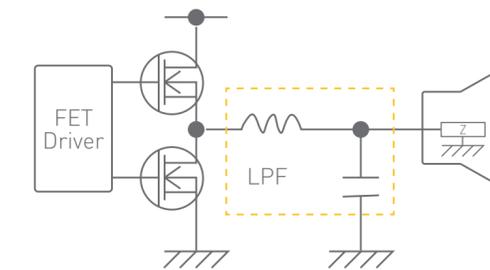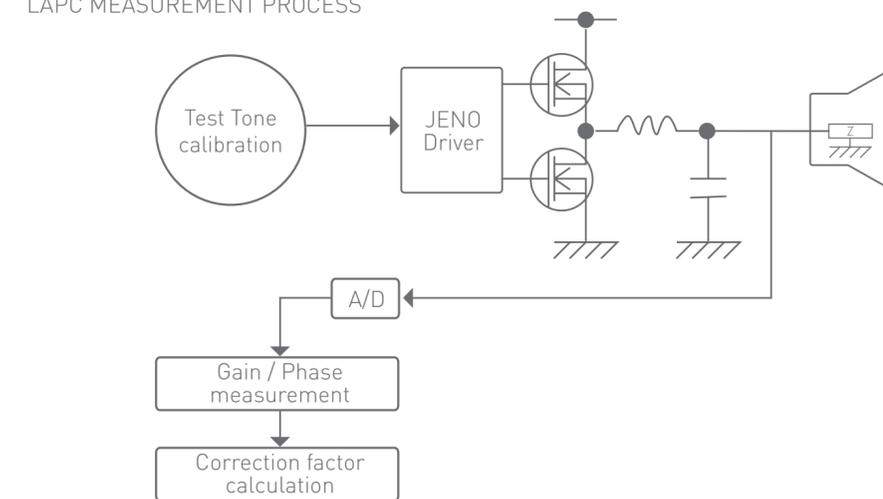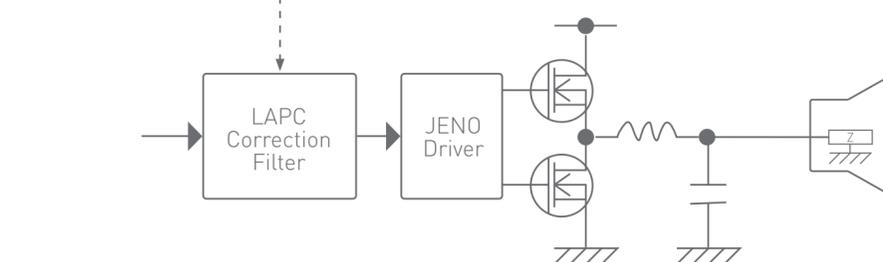
In this process, TSP (Time Stretch Pulse) is generated by the DSP for each input sampling frequency and output to the speaker. At the same time, the signal from the speaker terminal at that time is taken into the DSP. Inside the DSP, the gain/phase characteristics at the speaker terminal at that time are calculated from the difference between the captured signal and the generated signal, and the filter coefficient for cancelling the difference is calculated and stored in a non-volatile memory.

As shown in Fig.15-2, at the time of playback, the gain and phase characteristics matched to the connected speaker are given to the input audio signal by filtering with the DSP using the stored filter coefficient set. As shown in Fig.16, it is possible to eliminate the influence of speaker impedance and obtain a flat output with both gain and phase characteristics. In particular, flattening the phase cannot be achieved by NFB, which is a unique feature of Technics digital amplifiers.

Also, since the speaker impedance differs depending on the model, if the customer gets a new speaker, he can perform the LAPC calibration again to match the Technics amplifier easily with the new speaker.
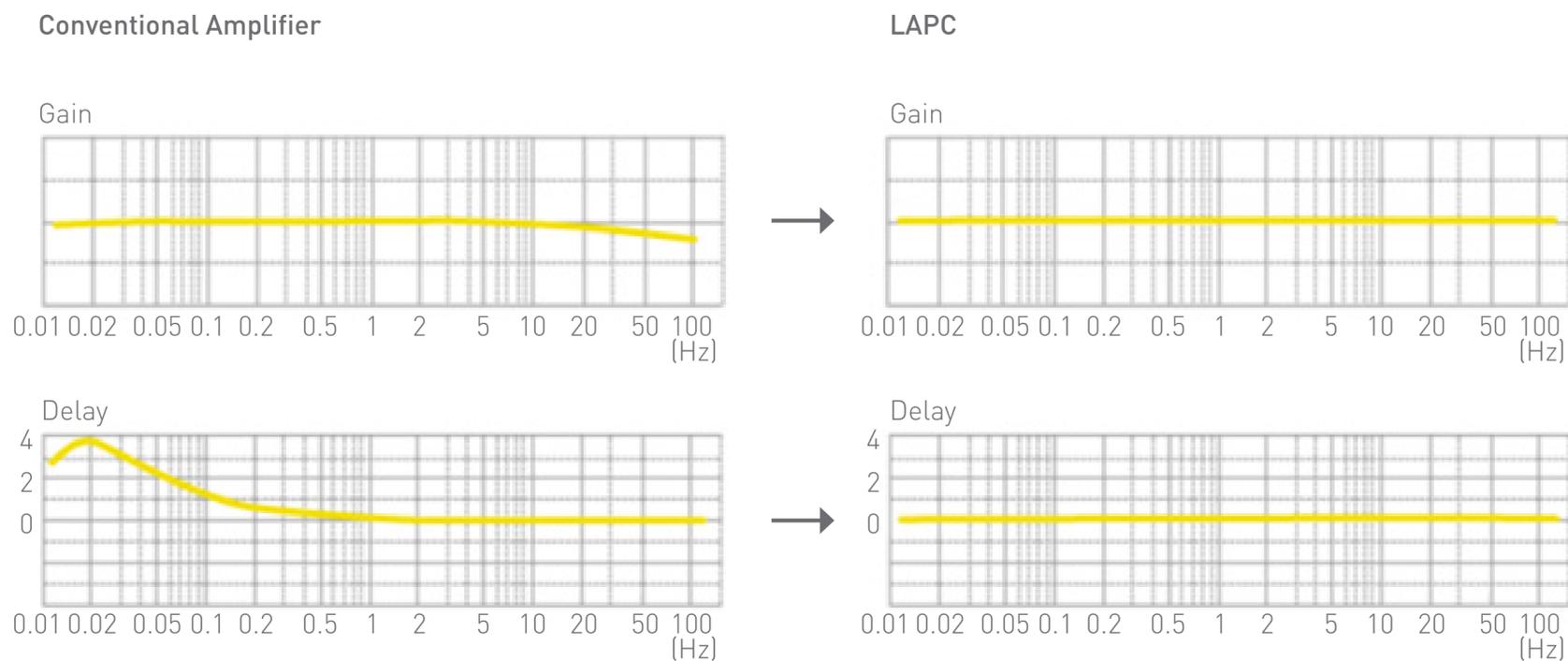
Conventional Amplifier

Gain

0.01 0.02 0.05 0.1 0.2   0.5   1   2   5 10 20   50 100
(Hz)

Delay

4
2
0

0.01 0.02 0.05 0.1 0.2   0.5   1   2   5 10 20   50 100
(Hz)

LAPC

Gain

0.01 0.02 0.05 0.1 0.2   0.5   1   2   5 10 20   50 100
(Hz)

Delay

4
2
0

0.01 0.02 0.05 0.1 0.2   0.5   1   2   5 10 20   50 100
(Hz)

Fig.16

7. Battery Driven Clock Generator

As described before, the accuracy of the clock is important for the digital amplifier, and the time accuracy affects the sound quality. The clock is generated by a crystal oscillator, and if noise is mixed in the oscillation circuit, jitter occurs in the oscillated clock. A constant voltage power supply is used for the stability of the oscillator circuit, but it is known that the constant voltage diode (Zener diode) used generates a minute noise that cannot be avoided by principle. Technics has adopted the "Battery Driven Clock Generator" to avoid such slight noise and obtain a clock with less jitter.

The operation is shown in Fig.17. By using two virtual batteries (electric double layer capacitor bank) and supplying power to the "Clock Generator" circuit from one charged battery, the other battery is alternately charged from the source "Power Supply". Thus, it becomes possible to separate the "Clock Generator" from the "Power Supply" and remove the noise that the "Power Supply" has. Fig.18 shows the circuit board mounted on SU-R1.

Fig.19 shows the effect. As shown here, an improvement of several dB was observed in the noise floor, which made it possible to achieve highly clear sound.
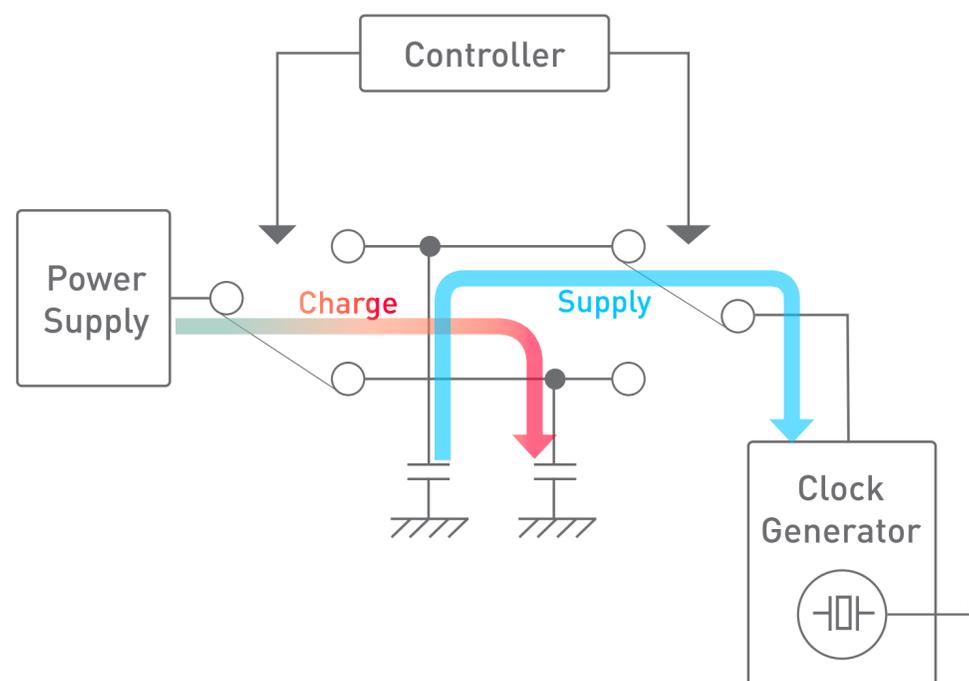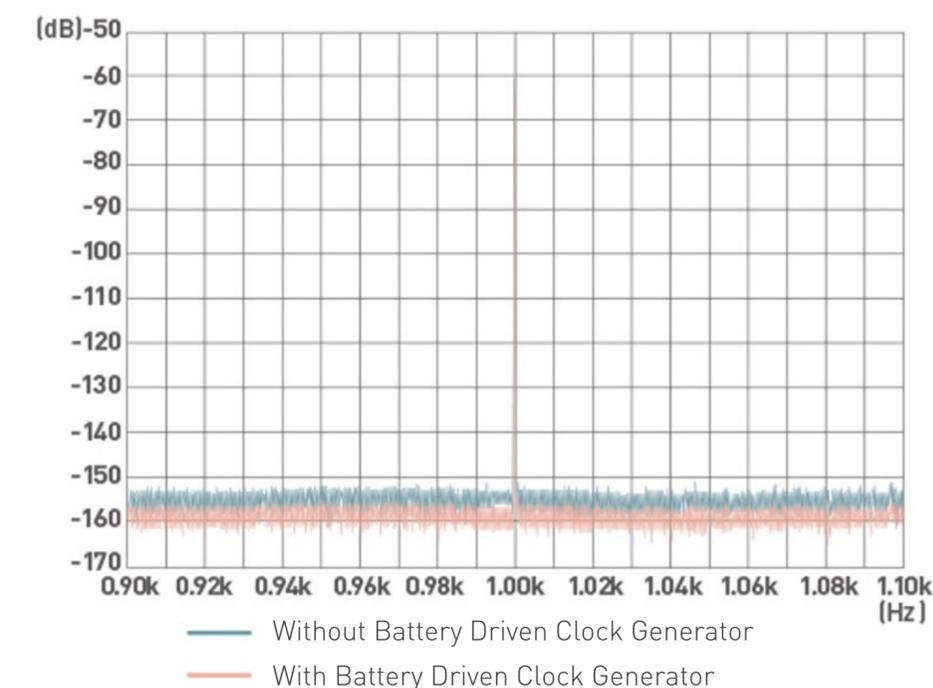

Fig.18 Circuit board mounted on SU-R1

Fig.17 Battery Driven Clock Generator

Fig.19 Spectrum of JENO output of 1KHz -60dB sinewave input

8. GaN

Fig.20

10 V/ div   20 ns/ div

Fig.21

Circuit Board Power Amp

GaN-FET

Fig.22

At the output stage of the digital amplifier, the PWM pulse generated by the "JENO" is turned ON and OFF (switching) by the FET driver, and power is supplied from the power supply to the speaker.

Since the PWM signal operates at high speed, the response of the output transistor also contributes significantly to the performance.

Normally, a power MOS-FET is used here, but gallium nitride (GaN) is used in high class machines such as SE-R1, SU-G30 and SU-R1000. Compared to MOS power transistors, GaN has an ideal characteristics of high-speed switching, low distortion, and low ON resistance, resulting in less loss, as shown in Fig.21.
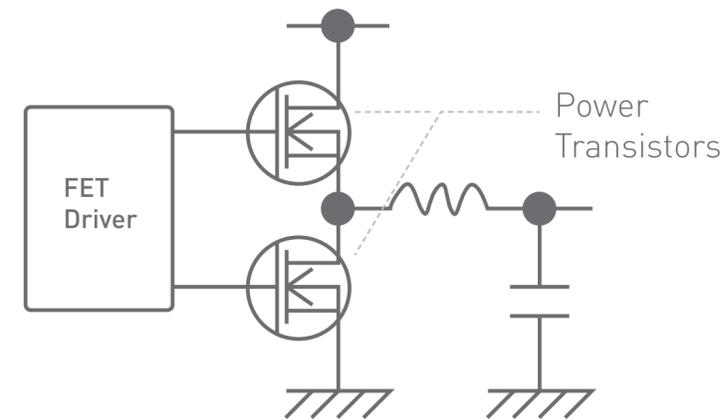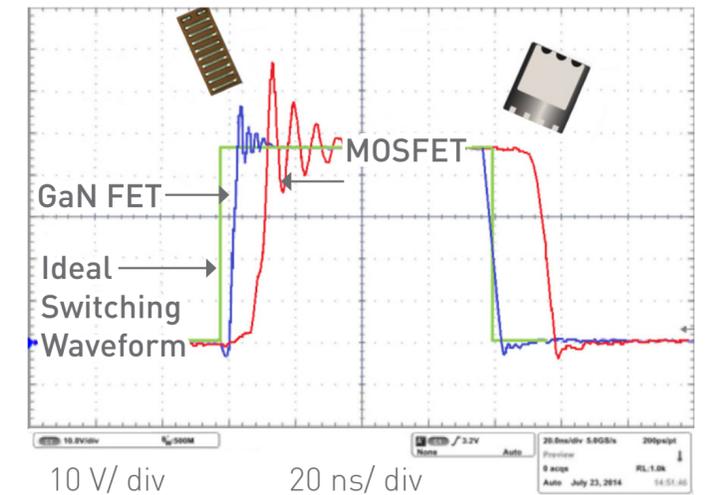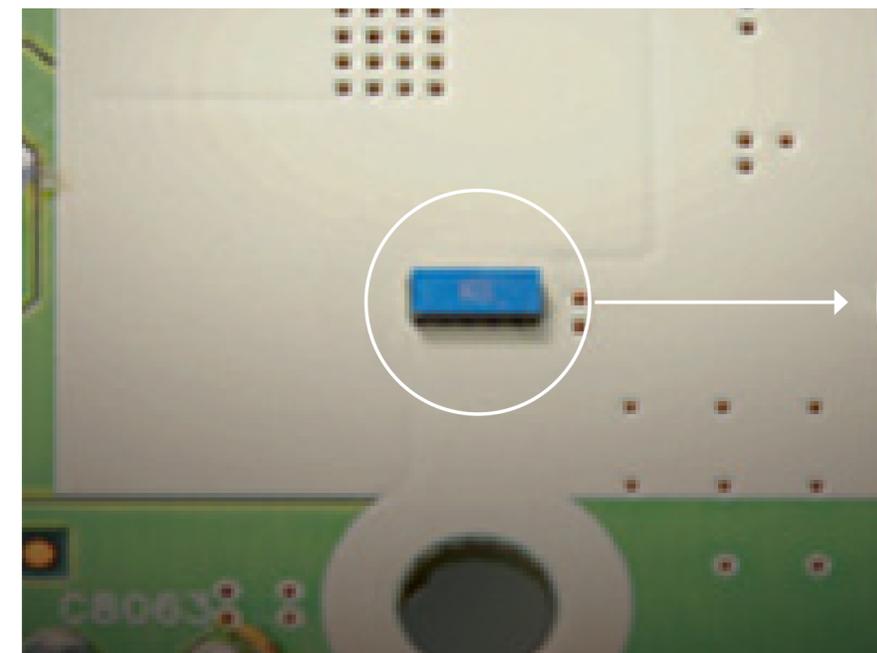
9. Conclusions

Technics Digital Amplifier high sound quality technology
is summarized above. In particular, "JENO" and "LAPC"
are the result of Technics's long years of R&D in digital
amplifiers, and we are proud to have achieved unique sound
quality enhancement technologies that no other company
can follow.

These technologies are implemented in Technics' SE-R1,
SU-C700, SU-G30, SU-G700, SU-R1000 and the "JENO"
custom LSI, which is further integrated into one chip, is
also applied to Technics Premium class integrated systems
which are on the market. Its sound quality is being highly
evaluated.

In the future, we plan to develop more technologies for
higher sound quality based on these.