

IC-7610

Technical Report

Volume 2



~ Introduction ~

In the IC-7610 Technical Manual Volume 1, we outlined the attraction of the IC-7610. In Volume 2, we will explain the RF direct sampling system used in the IC-7610 and RMDR which Icom has been using as the evaluation standard for receiver performance in recent years.

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What is the RF direct sampling system?

The RF direct sampling system is used in the IC-7610, following the IC-7300 and the IC-R8600 (the latter limited to 0.01~29.999999 MHz). We can imagine that someone who is interested in, or has doubts, may well ask "what is this RF direct sampling system?" Here, we will briefly explain it.

In short, as described in Volume 1, the received analog RF signal is converted to a digital signal in the A/D converter (ADC) and batch processed by the Field-programmable Gate Array (FPGA).

As you can see in the receiver block diagram, shown in Figure 01-01, the design from the antenna, through the bandpass filter (BPF), to the RF preamplifier is equivalent to a conventional (superheterodyne) radio. However, after the RF amplifier, there is an A/D converter instead of a mixer. Thus, as mentioned above, the RF signals are converted to a digital bit stream as is, without converting the RF signals to an intermediate frequency (IF). This is the reason why this architecture is called RF direct sampling.

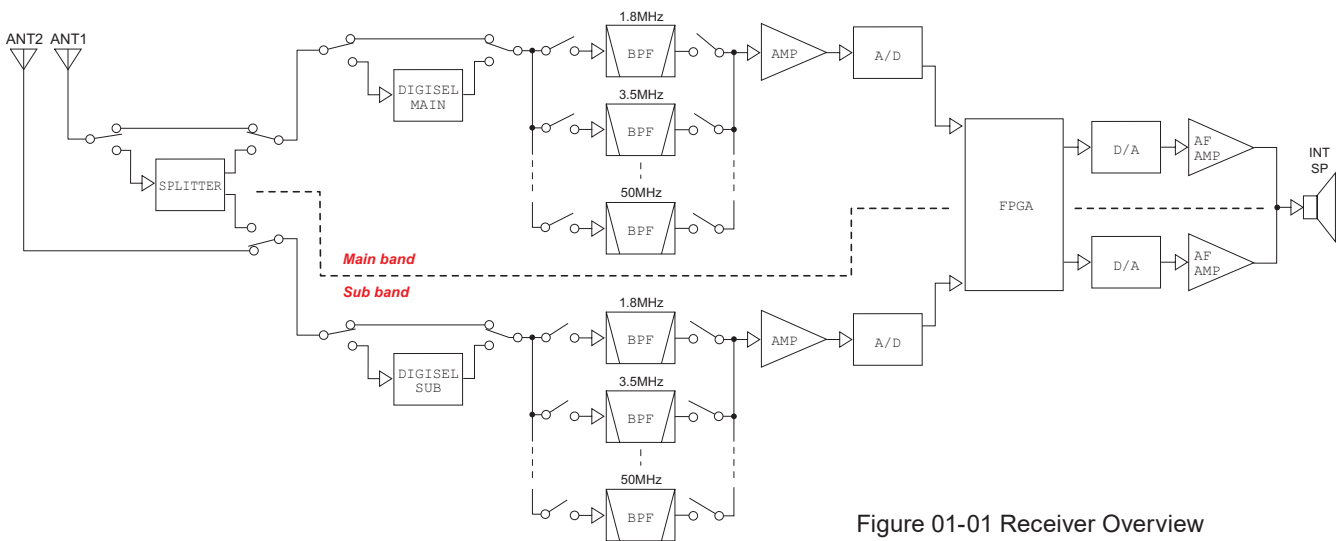


Figure 01-01 Receiver Overview

In addition, the reception method is described as "Direct Sampling Super Heterodyne Method" in the product brochure, and in Volume 1 of the Technical Report (published earlier). We will also briefly explain this point.

The reason for the terminology "direct sampling superheterodyne" is found in the architecture of the FPGA.

Figure 01-02 is a block diagram roughly describing the inside of the FPGA. The signal from the external A/D converter is input to the FPGA, and it is seen that it is input to a mixer (I/Q). This mixer (I/Q) converts the digitally converted RF signal to an I/Q signal. Since this conversion process also includes frequency conversion, there is a difference between analog and digital, but since it follows the superheterodyne principle, we are calling this architecture a "direct sampling superheterodyne system".

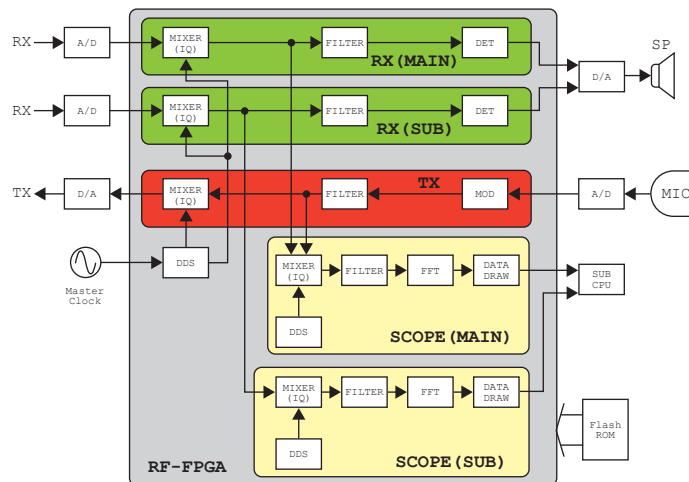


Figure 01-02 FPGA Internal Block Diagram

The advantage of the RF Direct Sampling system

About the direct conversion system

Let's talk about a general direct conversion system, before explaining the reason why Icom chose the RF direct sampling system.

The direct conversion system is a method of generating a signal equal to, or very close in frequency to the desired RF signal, with a local oscillator, and demodulating the RF signal by mixing it with the oscillator output. The number of parts used is small, and the design is very simple. For that reason, it is easy to make one chip offering miniaturization, low power consumption and low cost. Direct conversion is suitable for various applications, such as mobile phones.

However, an analog mixer is used for frequency conversion; it can present problems such as frequency drift in the local oscillator, spurious signals caused by local oscillator leakage, DC biasing of the mixer caused by a powerful interfering signal and second-order intermodulation distortion. Furthermore, other problems such as $1/f$ noise and I/Q mismatch (imbalance) can arise. It is also said that it is difficult to improve sensitivity, stability, and selectivity, termed the "3-S" of receivers, beyond certain limits.

There are other problems such as noise and I/Q mismatch (imbalance). It is also said that it is difficult to improve sensitivity, stability and selectivity, beyond certain limits. The superheterodyne system, widely adopted in high performance receivers, has largely overcome these problems.

Benefits of RF direct sampling method

We have explained that the superheterodyne method overcomes the problems of the direct conversion method, and the limits it sets to the "3-S" of receivers. So let us consider the RF direct sampling system.

The biggest merit of the RF direct sampling method is that all processing takes place in the digital domain. In the superheterodyne system, analog signals are processed in the frequency conversion (mixing) process. Since this frequency conversion uses the nonlinear distortion region of the analog element in the mixer, unnecessary nonlinear distortion occurs. On the other hand, with the RF direct sampling system, there is no analog mixer, as shown in figure 01-01 on the previous page. In other words, it is a circuit that generates no unnecessary nonlinear distortion.

Also, in analog signal processing, the noise from within and outside the radio equipment has various influences. By contrast, digital signals are not affected by noise at all. That is, there is no deterioration of the signal in the process after A/D conversion. If the signal processed inside the radio device does not deteriorate, the sound finally heard from speakers and headphones will be a faithful reproduction of the signal being received.

In the IC-7610, these digital signal processing (DSP) tasks are collectively performed within the FPGA, described later.

A/D conversion

A signal from the antenna is converted into a digital signal by an A/D converter after the target signal has been selected by the Digi-Sel (tracking preselector) circuit and a band pass filter (BPF: Band Pass Filter). So, how do you convert analog signals to digital signals? Here we briefly explain the principle.

First, as shown in Figure 01-03, we divide the analog signal into fixed time intervals (t seconds). Then, we read the value for each fixed time slot. (Fig. 01-04) This is called sampling.

Next, we convert the sampled value to an integer that can be handled with digital signals.

Next we map input values from a large, often continuous set, to output values into a countable smaller set.

This is called quantization. (In this case, processing is done by rounding and truncating the decimal part.)

After that, we convert the quantized value to a binary number. (Fig. 01-05) This is called encoding. As a result, when the analog signal shown in Fig. 01-03 is divided into 16 stages of t second periods and digital conversion with 4 bits, it is encoded as "1000111011111001001001011100101001001101000100010100011101101".

Generally, it is possible to digitize an analog signal faithfully by decreasing the sampling interval, and by increasing the number of levels when quantizing. On the other hand, as the amount of data increases, we need enough processing power to handle a considerable amount of data.

In general, Hertz (Hz) is used as a unit indicating the sample rate. 44.1 kHz is used in music CD's. This means that a 1 second sound is sampled by dividing it by 44,100.

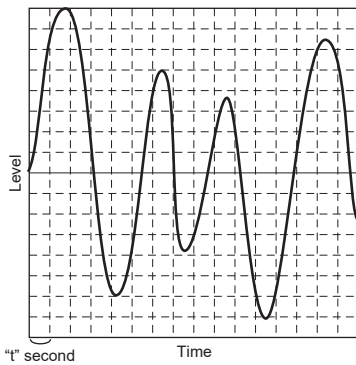


Figure 01-03

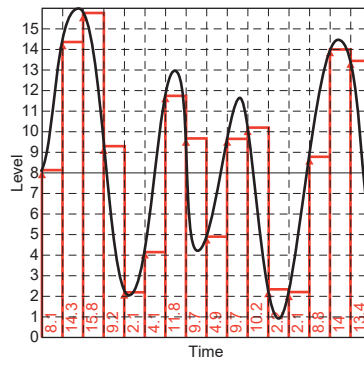


Figure 01-04

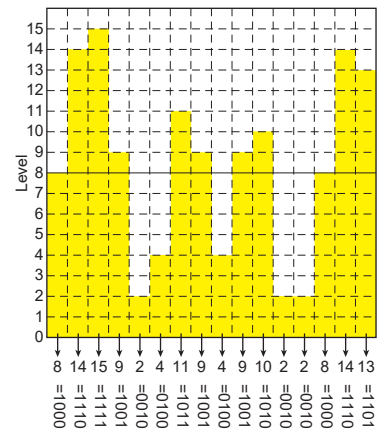


Figure 01-05

A/D conversion of the RF signal

To convert an analog RF signal to a digital signal, it is necessary to have a high-speed A/D converter capable of sampling at more than twice the frequency of the RF signal to be converted. This is stated in the sampling theorem.

What is the sampling theorem?

Assuming that the maximum frequency component of the signal band to be converted is F , a signal sampled at a frequency (F_s) higher than twice ($2 F$) thereof, removes high frequency components with a low pass filter (LPF: Low Pass Filter) so that the original signal can be completely restored.

Previously, a fast A/D converter capable of sampling RF signals was not practical. However, with the recent spread of mobile phones, smart phones, personal computer networks, and so on, the availability of devices that can sample at higher frequencies has increased. As a result, A/D converters capable of sampling the RF signal have become practical, so RF direct sampling has been realized.

Since the upper limit of the frequency that can be operated with the IC-7610 is 60 MHz, in order to sample the RF signal, an A/D converter capable of sampling at 120 MHz or more is required. For that reason, the IC-7610's A/D converter employs a device capable of sampling at 130 MHz.

FPGA internal processing

The signals digitally converted by the A/D converter are subjected to the following processing operations inside the FPGA.

- Conversion to I/Q signal
- Detection (demodulation)
- Signal processing such as noise reduction, noise blanker, selectivity filtering, NOTCH, PBT
- Scope signal generation
- Squelch control, and so on.

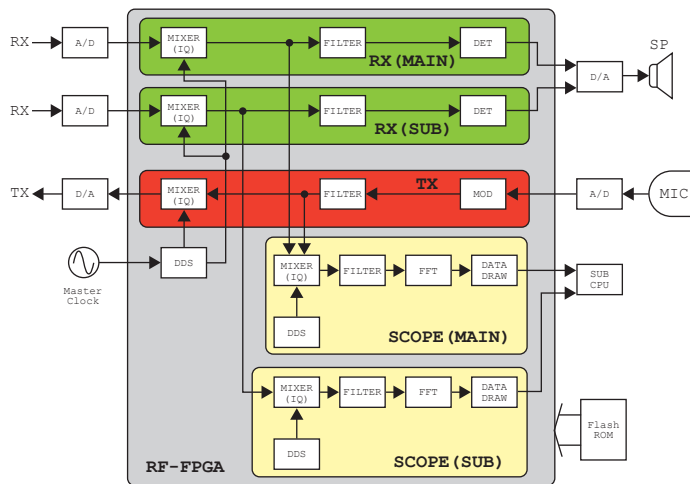


Fig 02-05 Comparison of LCDs

Conversion to I/Q signal and detection (I/Q = in-phase/quadrature)

The RF signal digitally converted by the A/D converter is converted to an I/Q signal in this mixer (I/Q) section. The converted I/Q signal is subjected to filtering and detection processing within the FPGA according to the operating mode selected on the IC-7610 main unit. If it is a voice mode (SSB, AM, FM), the I/Q signal is converted from analog to digital by a separate D/A converter, then it is output from the speaker or headphone through the audio amplifier circuit. If it is a digital data mode (RTTY, PSK), it is displayed on the display of the IC-7610, or the data is output from the USB 2 port on the rear panel, depending on the operating mode, after processing and decoding.

Also, demodulated sounds are output from the speaker and headphone jacks, for data as well as voice modes. This I/Q signal is scheduled to be output from the USB 3 port on the rear panel in a future firmware upgrade. (As of May 2018)

Scope signal generation

At the same time, the digitized received signal undergoes the FFT processing necessary for display of the spectrum scope, in the FPGA. The FFT-processed signal is sent to the CPU and displayed on the Scope screen.

Various signal processing features

Signal processing features such as Noise Reduction (NR), Noise Blanker (NB), NOTCH, PBT, and so on, which had formerly been implemented in the IF DSP, have now also been incorporated in the FPGA.

Basic of RMDR

What is RMDR?

RMDR = Reciprocal Mixing Dynamic Range. Reciprocal mixing in a receiver occurs when, during the reception of a wanted signal, a strong out-of-band interfering signal mixes with out-of-band skirt noise from the local oscillator (or A/D clock), producing mixing products that fall into the receiver's detection band, causing the receiver output signal-to-noise ratio to be degraded. RMDR is the ratio of the interfering signal level required to double the audio power output, minus the receiver's noise floor (in dB).

Reason for evaluating receiver performance with RMDR

Conventionally, it was common to express the performance of the receiver by the third order intercept point (IP3) value. However, when comparing receivers with excellent IP3 performance, even if the values are the same, there is a difference in actual operation. While investigating and studying the difference, we have focused on the RMDR that various institutions cite as one of the items for receiver performance evaluation.

At the same time, it has become established that RMDR performance is more important in high-performance receivers with IP3 specifications at a certain level. Therefore, when the IC-7850 and IC-7851 were released in 2014, RMDR was highlighted as a key performance parameter.

On the other hand, in a receiver using the RF direct sampling method, as mentioned above, there is no analog mixer circuit, so using IP3 as an index of performance does not make much sense. For these reasons, we at Icom have identified RMDR as an indicator of receiver performance, replacing IP3.

A case study where you can experience RMDR

Ham stations with local stations that transmit powerful radio signals, and when participating in contests with many stations on adjacent frequencies, have experienced that weak signals that are barely received by local stations are drowned out. Are many people not experiencing this? Receivers with excellent RMDR characteristics will be much less affected by these problems. Analyzing the sound output in this interference situation, and in situations that can be avoided by using the FFT scope, is shown in the figure below.

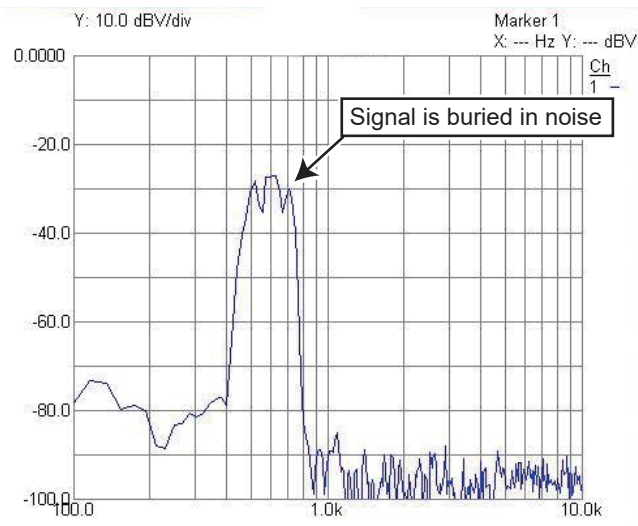


Figure 02-01 FFT analysis image of received voice output (Receiver with poor RMDR characteristics)

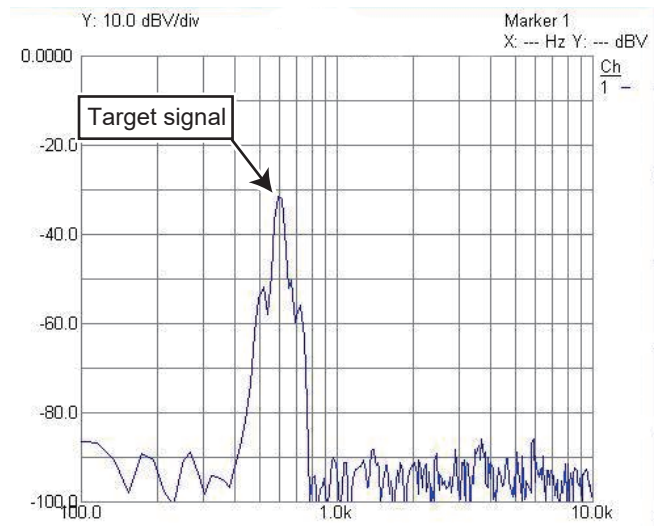


Figure 02-02 FFT analysis image of received voice output (Receiver with good RMDR characteristics)

The relationship between RMDR and phase noise

About Phase Noise

From here we will explain the basic points about RMDR.

RMDR indicates that the higher the value, the stronger the nearby interfering signal is. For this RMDR characteristic, the phase noise characteristic of the local oscillator signal is related to the conventional superheterodyne system. Phase noise is the noise component spreading like the foot of a mountain. It creates a noise pedestal above and below the fundamental frequency signal. This is illustrated in Figure 02-03.

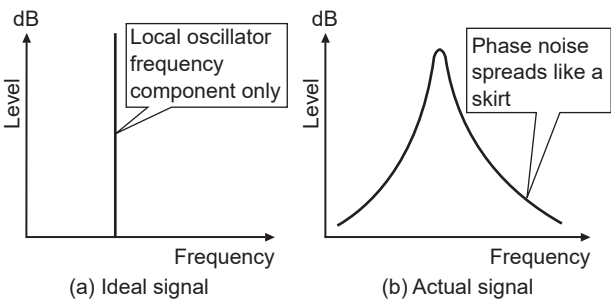


Figure 02-03 Ideal signal and actual signal

So why is the phase noise characteristic greatly related to the RMDR characteristic?

In a superheterodyne receiver with a conventional analog mixer circuit, two signals, the target signal and the LO signal, are input to the mixer circuit. If the phase noise characteristic of the input LO signal is poor, even if we try to receive a weak signal below the phase noise level, it will be drowned out by the phase noise itself, so the weak signal will not be recoverable from the mixer. On the other hand, suppression of phase noise means that the received signal will start to emerge from the noise, so it will become audible.

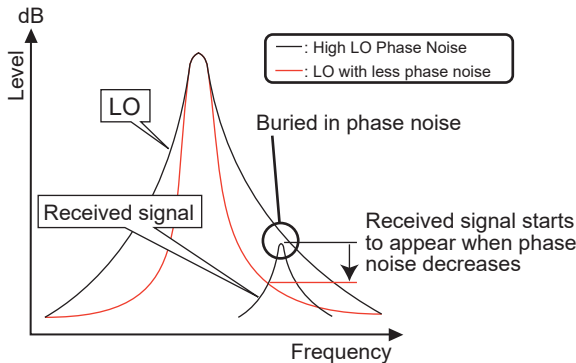


Figure 02-04 Phase noise and received signal

About Jitter

There is no analog mixer circuits in the IC-7610. Why does the same question arise as the conventional superheterodyne receiver? This is because, in RF direct sampling, the A/D converter and analog mixer circuit are considered to be equivalent, and phase noise characteristics and jitter are closely related.

Jitter is one of the items indicating the performance of the oscillator used for the reference frequency, the clock for each device, and so on. This is a deviation or fluctuation of the oscillator output signal in the time domain.

The clock signal necessary for sampling is applied to the A/D converter. If jitter is included in this clock signal, an error occurs in the sample value because the sampling point is shifted. This error acts as phase noise, and its noise spectrum forms "supersonic" components like the phase noise of the LO signal as encountered in the analog mixer circuit.

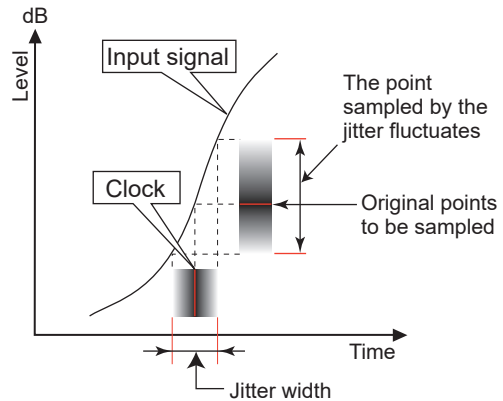


Figure 02-05 Jitter ~ Relationship between clock and sampling, as seen in the time axis

Under this condition of forming "supersonic" components, when weak target signals and strong nearby interfering signals are simultaneously input to the A/D converter, these three signals are "superimposed". Consequently, the conversion processing of a powerful adjacent interfering signal takes precedence and weak target signal cannot be processed. As a result, the RMDR characteristics of the receiver will be affected.

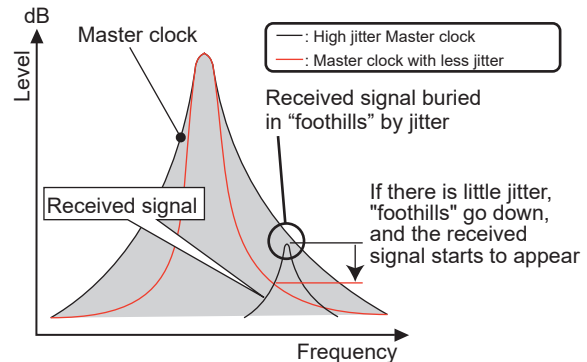


Figure 02-06 Jitter and received signal

Comparison with conventional transceivers

Comparison of RMDR

As shown in the brochure and product literature, the RMDR of the IC-7610 is about 110 dB. On the other hand, the RMDR of IC-7600 (a conventional transceiver) is about 80 dB, as explained in Volume 1, so here we have an amazing improvement of about 30 dB. Although the classes are different, even if comparing about 102 dB of the IC-7300, which pioneered RF direct sampling, with the conventional IC-7600 model, the benefit that RF direct sampling has on RMDR is tremendous, about 22 dB. This should be abundantly clear to the reader.

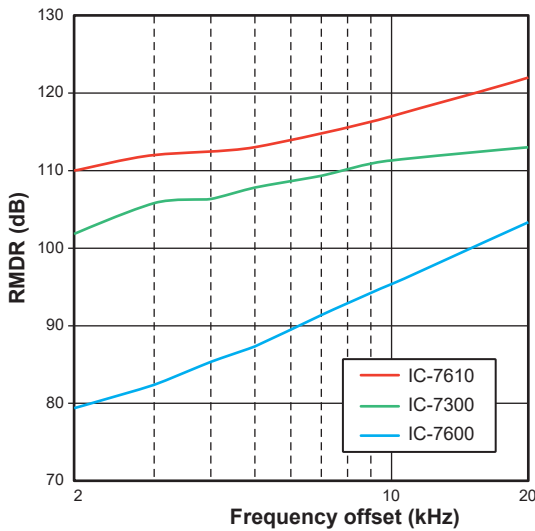


Figure 02-07 Comparison of RMDR

Comparison of transmission C/N and transmission phase noise characteristics

The low-jitter oscillator is the heart in the master clock in the IC-7610. As for the C/N characteristic of this master clock, there is no noise pedestal as seen with a conventional oscillator. The noise floor is also extremely low, and only the target signal can be seen.

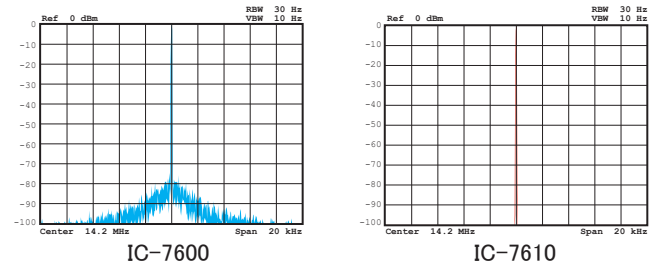


Figure 02-08 Transmit C/N Characteristics Comparison

In addition, the transmitted signal generated in the FPGA from the master clock with excellent C/N and phase noise characteristics will also have excellent C/N characteristics comparable to the receiver, and it is possible to obtain an extremely clean transmitted signal with minimized phase noise.

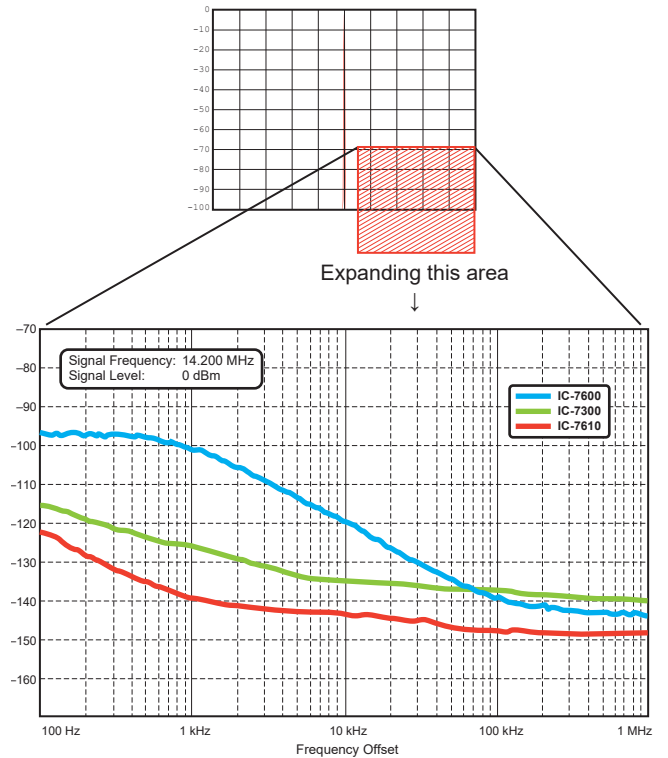
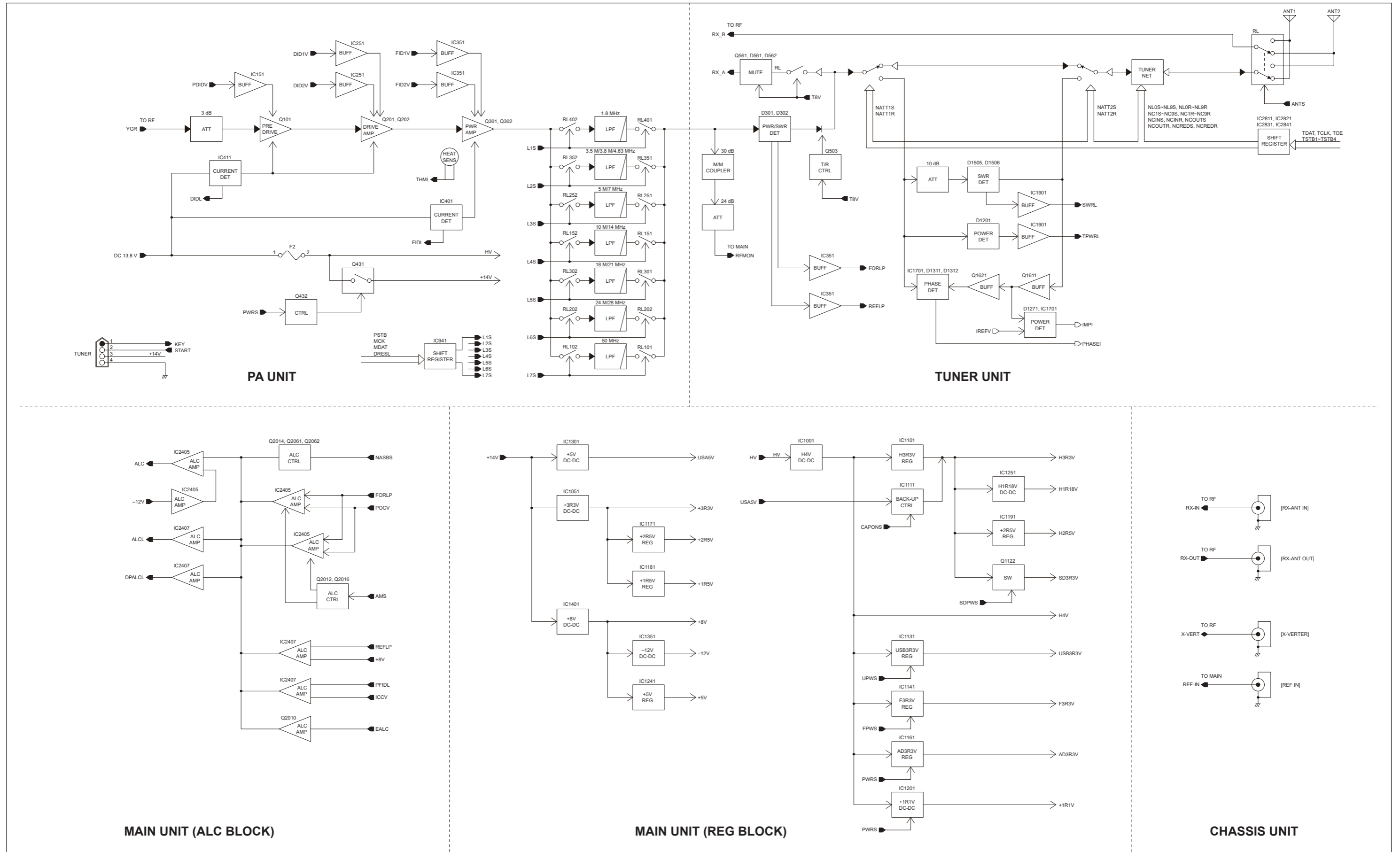
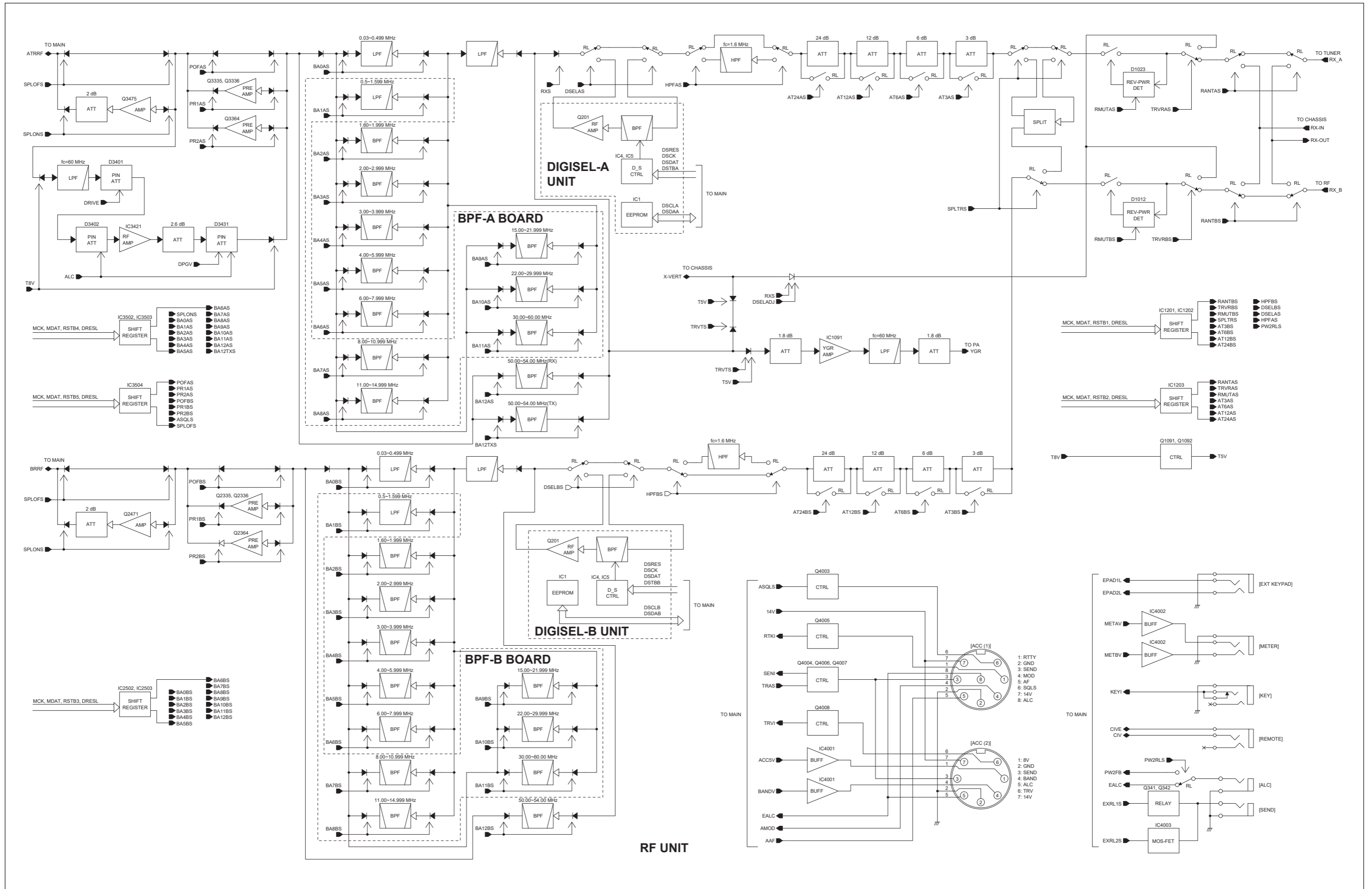
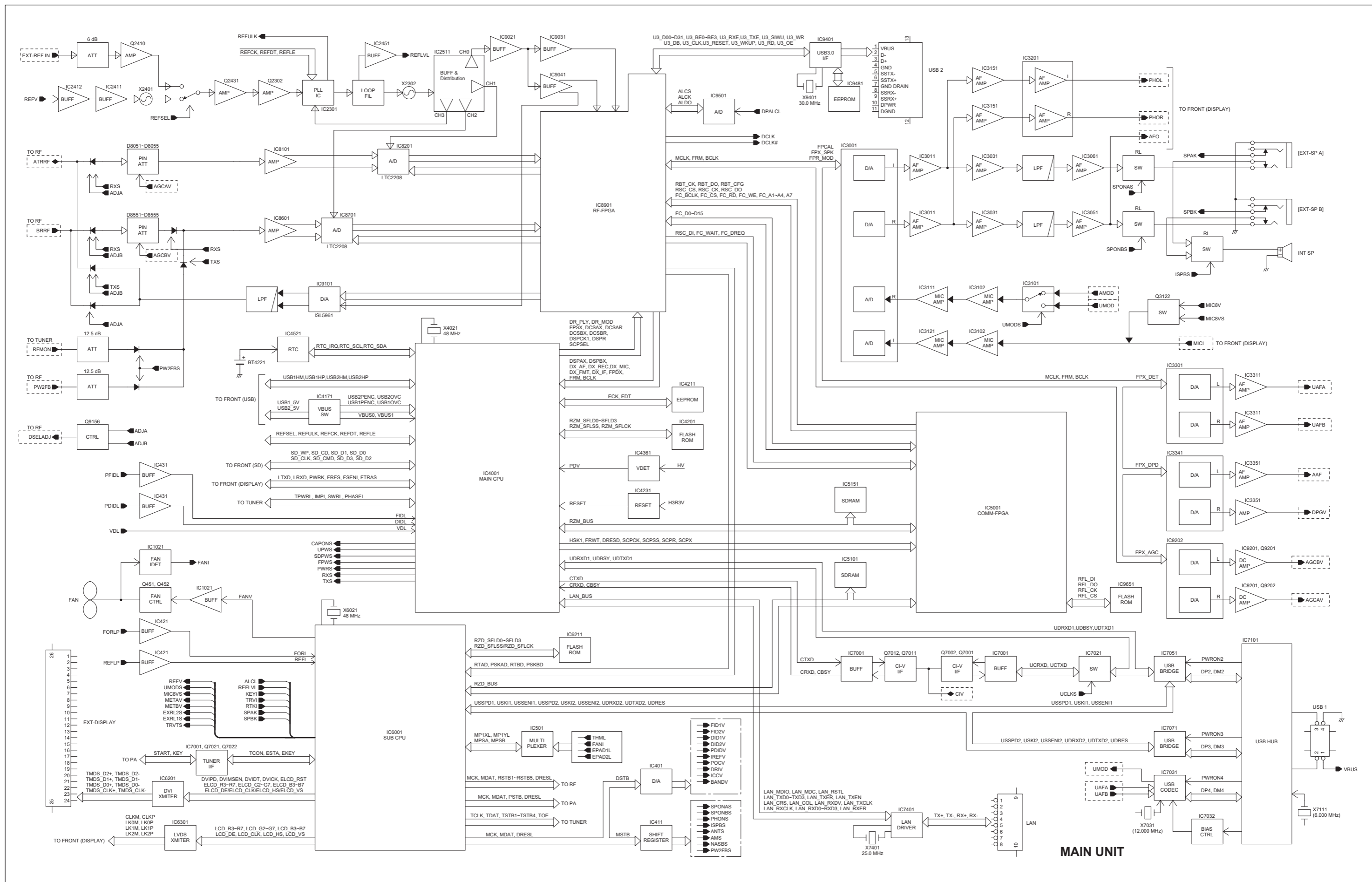
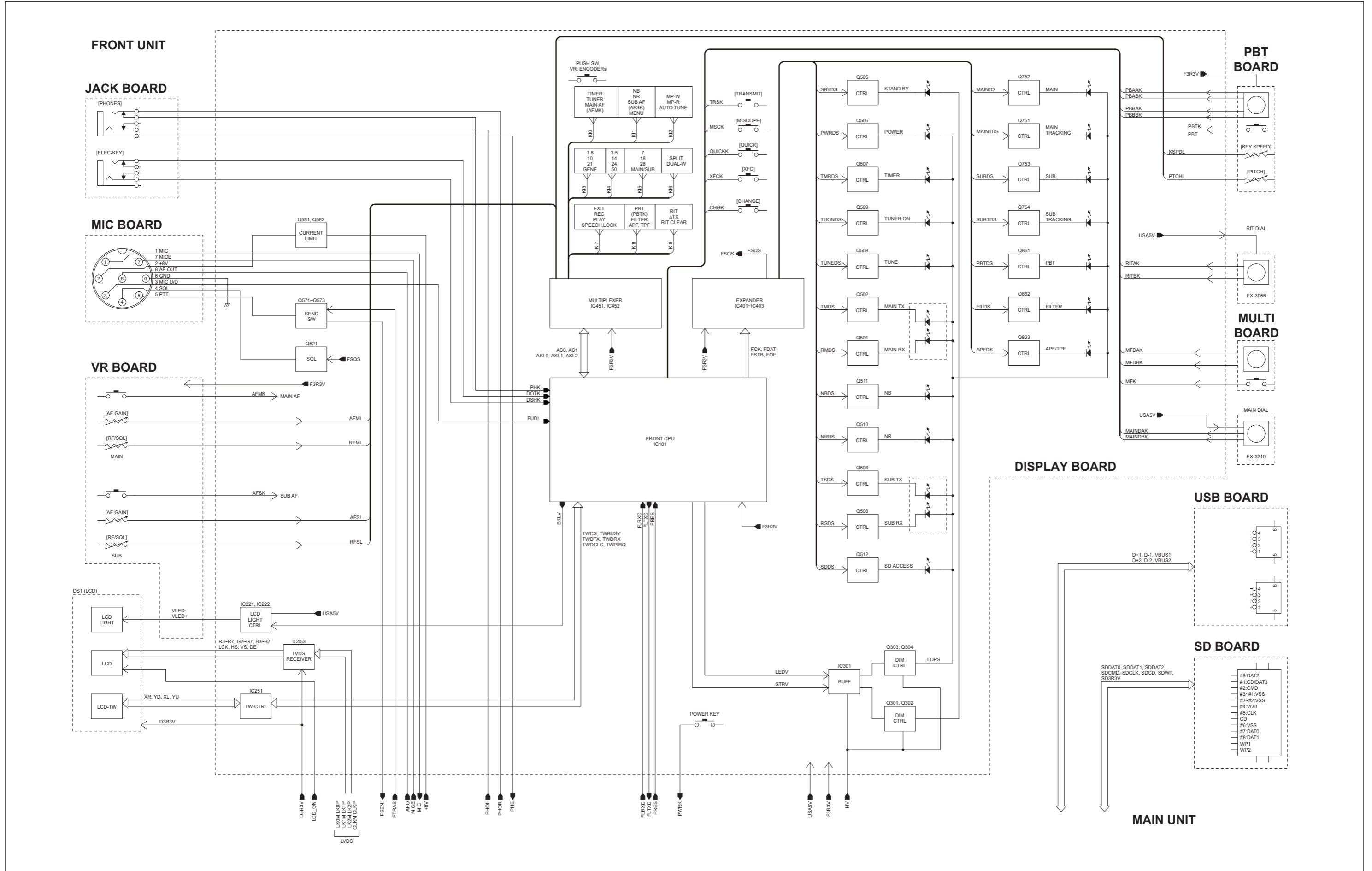


Figure 02-09 Transmit Phase Noise Characteristics Comparison









Count on us!

