



# IC-756 PRO II

# TECHNICAL REPORT



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The IC-756PROII is the high performance HF transceiver of choice for today's discriminating amateur radio operator. Icom's engineers took cutting-edge digital technology and paired it with Icom's extensive experience with analog technology. The result is a major advancement of Icom's original digital IF filter which, in the earlier IC-756PRO model, enjoyed a great reputation around the world.

The IC-756PROII uses the same 32-bit floating point DSP and a 24-bit A/D-D/A converter as the IC-756PRO. It is now possible to execute the digital IF filter, noise reduction and the digital IF filter in the AGC loop processing, and to select the soft/sharp filter shapes. The IC-756PROII employs exclusive DSP/analog circuit matching to further improve receiver performance.

Icom's engineers analyzed the influence of the AGC loop upon the received audio, matching it to an analog circuit suitable for the dynamic range of the A/D converter and the other parts used, and also re-examined the core stage of the receiver (ranging from RF top to mixer circuit), to distribute the mixer levels properly. As a result, the matching of digital and analog technology has attained a level never before achieved.

This technical report does not explain in depth all the digital engineering with its many calculations and formulas. Instead, it focuses on the DSP engineering in an easy-to-understand manner. This report also explains why the 32-bit floating point DSP and 24-bit A/D-D/A converter are included. The dynamic range of the 32-bit floating point DSP and the 24-bit A/D-D/A converter may seem to be an over specification for amateur radio. But this is not the case. This technical report helps clarify these points.

It is Icom's hope that in providing you with this report you will discover the IC-756PROII's many digital advantages. Enjoy!

## 2. Features

### 32-bit floating point DSP and 24-bit A/D-D/A converter

The adoption of a 32-bit floating point DSP and 24-bit A/D-D/A converter in the IF stage (36kHz) was originally developed by Icom. It enables various digital functions which amateur radio operator's desire.

#### ■ 51 types of digital IF filtering

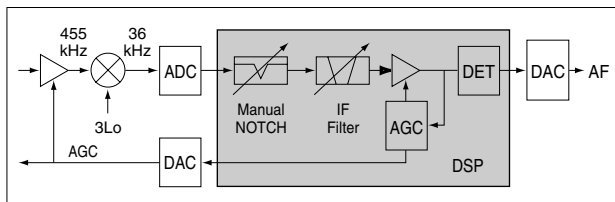
The digital IF filter has superior filtering performance and a distinguished shaping factor that demonstrates the power of 32-bit floating point DSP. The digital filter is completely free from deterioration due to deviations in band characteristics, temperature change, or mechanical vibration, all of which have been observed in analog filters. It also provides excellent ripple characteristics that have never been available with analog filters. The passband (50Hz–3.6kHz) of the digital IF filters used for IC-756PROII come in 51 types. This function allows 3 of these 51 types to be pre-set for each mode and to be changed instantaneously by using the filter button, depending on the situation.

#### ■ 2 types of filter shape (Sharp/Soft)

Changing the IF filter shape is a feature that is not available with analog radios. Select the filter shape from two types, Sharp and Soft, depending on the purpose, operating band, band conditions, etc. It is possible to set CW and SSB filter characteristics independently and also select the filter shape while actually receiving a signal.

#### ■ Digital IF filter in AGC loop

The digital IF filter, manual notch filter, etc. are located in the AGC loop, using DSP, which completely eliminates interference from adjacent strong signals. This allows the AGC to be operated only on the selected frequency. It is also possible to pre-set the operation of AGC in each mode in accordance with 13 types of time constants.



#### ■ Digital twin PBT

The IC-756PROII is equipped with DSP based twin PBT filtering. It provides excellent performance on a completely different level than analog filters. Set the frequency, and then adjust the passband width of the received signal in steps of 50Hz using the dedicated two-position knob. The passband width and direction of shift may be graphically displayed on the LCD, if the operator chooses.

#### ■ High-accuracy digital modulation and demodulation in all modes

The DSP unit allows you to increase transmit/receive audio levels, modulation, and demodulation — even while decoding RTTY. This makes it possible to set the passband width of the IF filter for SSB transmit to 3 different stages. The DSP unit also provides a demodulation level suitable for high-grade HF performance and high-fidelity sound.

#### ■ Manual notch with superior attenuation level

The IC-756PROII's manual notch filter has extremely sharp characteristics for processing in the DSP and provides tremendous performance for attenuation levels >70dB. Analog notch circuits are susceptible to fluctuations in attenuation or changes in temperature. The DSP-based manual notch provides stable performance and is not susceptible to such changes. Also, the DSP signal processing executed within the AGC loop completely shuts off undesirable signals, even with the AGC set to high speed. An automatic notch is included to further enhance receiver performance.

#### ■ Demodulator/decoder for RTTY

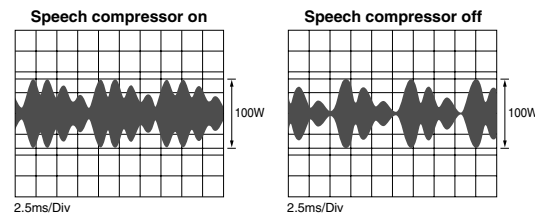
This transceiver is equipped with a demodulator and a decoder for BAUDOT RTTY as a standard feature. On-air station calls may be recognized instantaneously by reading the received RTTY message directly on the IC-756PROII's LCD — no personal computer or external components are required. The transceiver is fitted with an on-screen tuning indicator that allows the RTTY to be fine tuned with ease. A DSP based twin-peak audio filter further improves the S/N ratio. This filter will reduce interference that appears between each tone (mark and space), which cannot be removed by conventional analog filters. This twin-peak audio filter works to capture noise-level signals accurately and to significantly reduce the generation of noise distortion.



RTTY reception screen

#### ■ New-generation speech compressor

The DSP based speech compressor enhances the readability of your transmitted signal at a receiving station without any distortion, even when the compression is set to a high level. The gradation of voice processing is extremely close to the original sound. This assures superior sound quality at all compression levels.



#### ■ Microphone equalizer (enables 121 different settings)

The IC-756PROII is equipped with a microphone equalizer that makes it possible to set the frequency characteristics of the transmitted signal in 11 different stages for both the high-tone range and the low-tone range. Considering all permutations, this provides for a total of 121 different settings. With this flexibility of DSP based waveform shaping, it is possible to adjust transmit audio quality depending on the application. For example, it is possible to set the dynamic sound quality for "Pileup" or to set pleasant sound for "Ragchewing".

#### ■ Variable level type noise reduction

The 32-bit floating point DSP has excellent calculation performance, which processes complex and sophisticated algorithms. This allows the DSP to attenuate noise without delay and extracts noise-level signals. It is possible to vary the suppression level in 16 stages.

#### ■ CW keying waveform shaping function

DSP controls the rise and fall of the CW transmit waveform. The result is a proper CW waveform. The rise/fall timing is selectable to 4 stages of 2ms, 4ms, 6ms and 8ms. This makes it possible to set a "Soft" or "Hard" CW signal, depending on your preference.

## Enhanced functions

### ■ Advanced receive functions

The RF stage's front-end receive mixer is designed in a 4-element configuration. In the IC-756PROII, this configuration is used in the BPF stage at the RF top. Each element is examined to tune the circuit after RF stage to mixer, which makes it possible to enhance the receive performance. This significantly reduces 3rd and other order distortion and provides a wide dynamic range. This means the IC-756PROII will accurately capture weak signals that analog type radios cannot hear, even in low bands with high noise levels.

### ■ Real-time spectrum scope

A real-time spectrum scope is recognized as indispensable for DX hunting. The IC-756PROII's spectrum scope uses two colors to display all RF signal activity within a user-selectable bandwidth. One color indicates real-time RF signal activity, while the other color provides peak hold indication. The spectrum scope may be used for sophisticated applications such as identifying the band conditions, quick discovery of stations, and confirmation of interval or call-back frequency. Additionally, you may monitor normal band conditions, while you display sub readout or transmit markers. In case of high-band noise, the IC-756PROII is equipped with an attenuator (10/20/30dB) dedicated to the spectrum scope, which allows a reduction of total signal level at the band scope without affecting the received signal.



Real-time spectrum scope screen

### ■ Exceptionally clear SSB transmit signal

Using Icom's advanced digital PSN modulation, the IC-756PROII emits high-quality signals, which makes its transmitter suitable for use as the exciter of a linear amplifier. Unwanted sidebands and carrier leaks are almost completely eliminated. Further, the transmitter employs a wide band power amplifier that incorporates highly reliable bipolar transistors (2SC5125 × 2). The linearity and IMD characteristics achieve superior signal quality never before seen in any amateur radio transceiver. This makes it possible to transmit RF signals with significantly reduced distortion.

### ■ Built-in high-stability reference crystal oscillator

The IC-756PROII's transceiver exhibits excellent frequency stability of < 0.5ppm. This assures stable communication even for RTTY and SSTV modes for which particularly high frequency stability is required.



High-stability reference crystal oscillation unit

## The best in operating convenience and features

### ■ Dual-watch

Dual-watch enables simultaneous two-frequency receive in the same band, providing identical band and filter configurations in both receive systems. This makes it possible to receive two signals simultaneously as if two separate receivers are being used. This greatly enhances split frequency operation; enjoy enhanced DX-operation by searching for pickup frequencies while watching the transmit frequency of a DX station experiencing pile-up. Or have a QSO while simultaneously monitoring a DX net.

### ■ Triple band stacking register

With the push of a band button, get quick memory recall of three preferred operating settings (including antenna port) per band. Band or mode hopping has never been easier. It's the ultimate in multi-mode flexibility.

### ■ Digital Voice Recorder (DVR)

The DVR feature is an indispensable function for DX hunting and contests. The IC-756PROII is equipped with a DVR with 4 channels for transmit and 4 channels for receive, for a total of 8 channels. High quality digital mapping of the transmitted or received analog signal provides high quality audio reproduction, resulting in a natural sounding voice without any noticeable degradation. It is also possible to use these 4 communication channels by allotting them freely with a total recording time of 90 seconds. Each of the 4 channels for receive has a recording time of 15 seconds, or 60 seconds total. Press the key once in any TFT display mode and it becomes possible to not only record or reproduce voice but also to record for up to 30 minutes continuously. The receive audio may be reproduced for the most recent 15 seconds back to an interruption in recording. By constructing the simplified control unit (page 26) and connecting it to the microphone connector, digital voice recorder function may also be operated.



DVR key

### ■ Full-scale electronic keyer

Plug a CW, iambic paddle into the electronic keyer jack on the front panel. Especially handy during long hours of operation, it is possible to set the CW speed between 7 and 56WPM. The discriminating operator may also set the dot/dash keying ratio (2.8:1 to 4.5:1) and polarity, depending on preference. The keyer may also be set for either right or left hand use. For the CW operator who prefers not to use the IC-756PROII's built-in electronic keyer, an ordinary key jack is available on the rear panel, for bug or straight key and is fully compatible with external keyers or PC keying.

### ■ Multi-function memory keyer

Enhance your contest operations. The IC-756PROII is fully equipped with a convenient memory keyer, offering features such as memory content editing function, auto-repeat function, serial contest number automatic count-up function, contest number abbreviating function, and more. These features will reduce effort when repeating a formatted contents for calling CQ, continuous transmission of call sign, or contests. Since it is possible to confirm the contents of memory on the display, transmission mistakes are eliminated. Construct the simplified control unit (page 26) and connect it to the microphone connector to enhance operation of these memory keyer functions.

### ■ Quick split function

When the split button is pressed and held, the frequency of the sub-VFO is adjusted to the frequency in the main VFO. Using the split function, it is also possible to control the following:

1. Vary the transmit frequency via the main dial.
2. Direct entry of the designated frequency.
3. Direct entry of the shift frequency.

You are now ready to “bag the DX” while other operators are still tuning up.

### ■ Preamp and attenuator

The IC-756PROII incorporates two types of receive preamplifiers: Preamp 1 (10dB) emphasizes modulation across all bands, and preamp 2 (16dB) emphasizes sensitivity especially for high bands. The attenuator is selectable in three stages, 6, 12 and 18dB. When there is a strong signal from a local commercial station it becomes possible to control the generation of distortion at the RF stage of the receiver. It is also possible to retain the preamp and attenuator settings for each band.

### ■ Variable noise blanker

The transceiver uses a new noise blanker design that provides significant reduction of pulse-type noise. The noise blanker also greatly enhances weak signal copy, allowing the operator to change the sensitivity in 100 stages in accordance with the noise level without distorting the target signal.

### ■ Frequency shift function for change from/to SSB to/from CW

A frequency shift function automatically adjusts the CW carrier point when selecting from SSB mode, or vice versa. You may select “Shift function off” whereby the frequency remains the same (by moving the carrier point), or “Shift function on” in which the frequency is shifted without moving the carrier point. Using CW-R mode it is possible to set the carrier point to USB.

## Enhanced TFT color display

### ■ High visibility

A high visibility 5-inch TFT color display has been integrated into the IC-756PROII to provide ease of use and clear indication of the radio's many features. Various function settings such as frequency, memory frequency, comment, filter setting status, RTTY tuning indicator, and more are displayed in the upper portion of the display. The lower portion of the display gives voice memory, characters of received RTTY, and the real-time spectrum scope information. The display color may be selected from 8 types, from vivid color to muted grays. 7 different font types may also be selected. These settings may be made in any combination – customize your display to best suit your personality or favored operating set-up.

### ■ Memory channel/memory list

The transceiver is equipped with 99 regular memory channels and 2 scan edges, totaling 101 channels. It is possible to enter text of up to 10 characters in each memory channel. It is also possible to display a list of up to 13 memories at a time.

### ■ Simplified set mode

The IC-756PROII has a list display that allows the status of each set mode item to be seen at a glance. Each function is divided into 4 setting groups and multiple items are listed or displayed to allow quick access to the desired item. This allows the many functions of the radio to be used with ease.

### ■ Digital meter simultaneously displays 4 transmit level indicators

With the digital meter (including peak-hold), it is possible to confirm the output power, ALC, SWR, and COMP, all at the same time while transmitting. The signal strength is also displayed while receiving.

## Enhanced functions

### ■ Antenna system

- High-speed built-in auto antenna tuner covering up to the 50MHz band
- 2-piece antenna terminal (incorporated with auto antenna selector)
- Dedicated receive antenna connector

### ■ Receive system

- General coverage receive (30kHz to 60MHz)
- Control of RF gain and squelch with one knob

### ■ Transmit system

- IF Monitor function allows the transmissions of your station to be listened to locally
- Built-in 50 frequencies of tone encoder/decoder
- VOX function allows the automatic selection of transmit and receive for “hands free” operation
- All-mode power control function

### ■ CW system

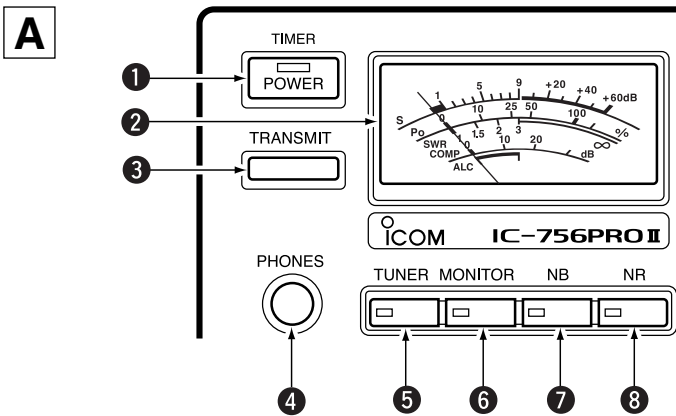
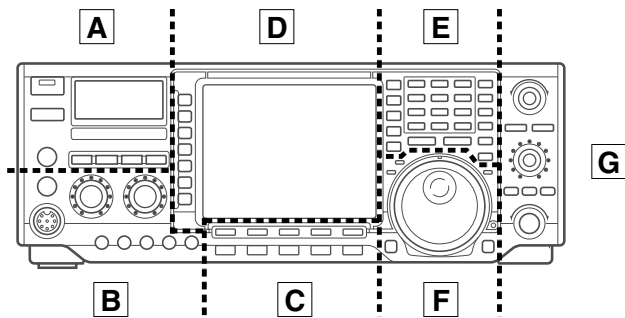
- CW pitch control function allows the CW receive tone to be set to a desired frequency (300 to 900Hz continuously)
- Double key jack allows 2 types of keys to be connected simultaneously
- Full break-in function allows receive during a break while keying

### ■ Operation system

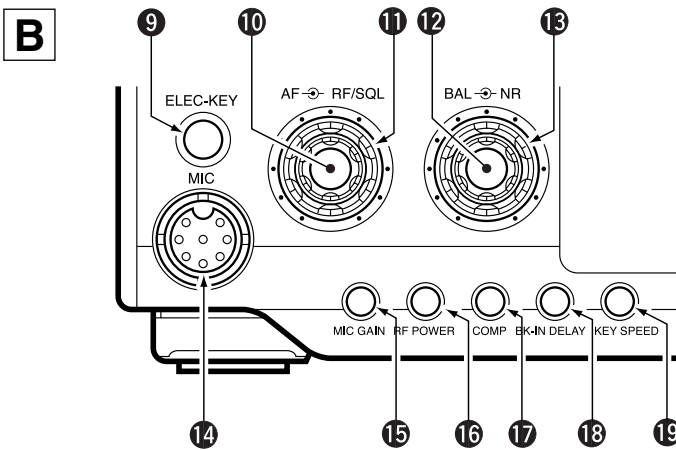
- 5-channel memo pad saves frequency and mode (It is also possible to change the 5-channel memo pad to a 10-channel type.)
- RIT and TX variable up to  $\pm 9.999$ kHz
- 1Hz pitch tuning
- Optional frequency speech allows the S-meter level to be announced
- High visibility needle type white-tone analog meter
- Various scanning functions (program, memory, select memory,  $\Delta F$ )
- Auto Tuning Step
- Dial-lock
- Split frequency lock
- Torque adjustment mechanism for main dial
- Band edge beep function
- CI-V terminal allows control from a personal computer
- Clock/timer function
- AH-4 control circuit

### 3. Front and rear panel

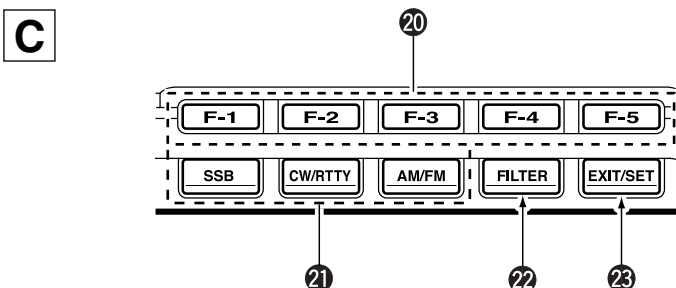
#### 3-1 Front panel



- ① POWER SWITCH [POWER TIMER]
- ② S/RF METER
- ③ TRANSMIT SWITCH [TRANSMIT]
- ④ HEADPHONE JACK [PHONES]
- ⑤ ANTENNA TUNER SWITCH [TUNER]
- ⑥ MONITOR SWITCH [MONITOR]
- ⑦ NOISE BLANKER SWITCH [NB]
- ⑧ NOISE REDUCTION SWITCH [NR]

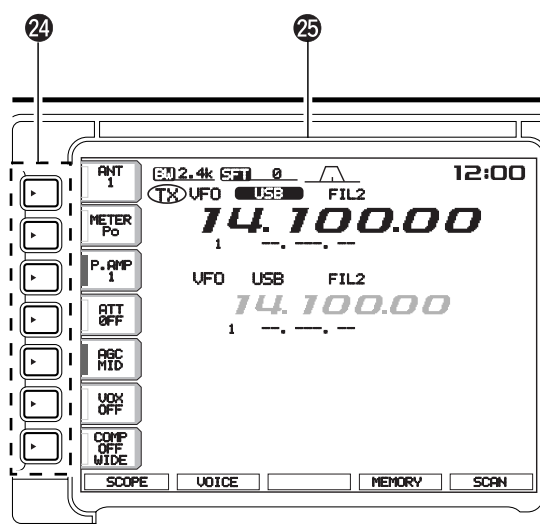


- ⑨ ELECTRONIC KEYSER JACK [ELEC-KEY]
- ⑩ AF CONTROL [AF]
- ⑪ RF GAIN CONTROL/SQUELCH CONTROL [RF/SQL]
- ⑫ BALANCE CONTROL [BAL]
- ⑬ NOISE REDUCTION LEVEL CONTROL [NR]
- ⑭ MICROPHONE CONNECTOR [MIC]
- ⑮ MIC GAIN CONTROL [MIC GAIN]
- ⑯ RF POWER CONTROL [RF POWER]
- ⑰ COMPRESSION LEVEL CONTROL [COMP]
- ⑱ SEMI BREAK-IN DELAY CONTROL [BK-IN DELAY]
- ⑲ ELECTRONIC CW KEYSER SPEED CONTROL [KEY SPEED]



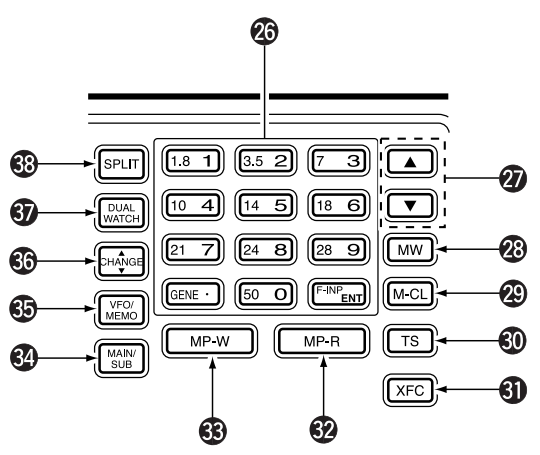
- ⑳ LCD FUNCTION SWITCHES [F1]-[F5]
- ㉑ MODE SWITCHES
- ㉒ FILTER SWITCH [FILTER]
- ㉓ EXIT/SET SWITCH [EXIT/SET]

**D**



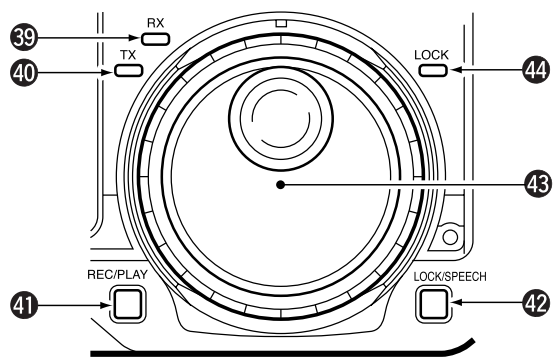
- 24 MULTI-FUNCTION SWITCH GUIDE
- 25 LCD FUNCTION DISPLAY

**E**



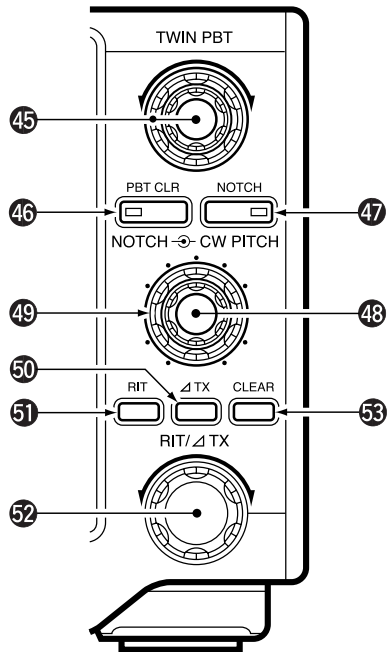
- 26 KEYPAD
- 27 MEMORY UP/DOWN SWITCHES [▲][▼]
- 28 MEMORY WRITE SWITCH [MW]
- 29 MEMORY CLEAR SWITCH [M-CL]
- 30 QUICK TUNING SWITCH [TS]
- 31 TRANSMIT FREQUENCY CHECK SWITCH [XFC]
- 32 MEMO PAD-READ SWITCH [MP-R]
- 33 MEMO PAD-WRITE SWITCH [MP-W]
- 34 MAIN/SUB CHANGE SWITCH [CHANGE]
- 35 VFO/MEMORY SWITCH [VFO/MEMO]
- 36 MAIN/SUB CHANGE SWITCH [CHANGE]
- 37 DUALWATCH SWITCH [DUALWATCH]
- 38 SPLIT SWITCH [SPLIT]

**F**



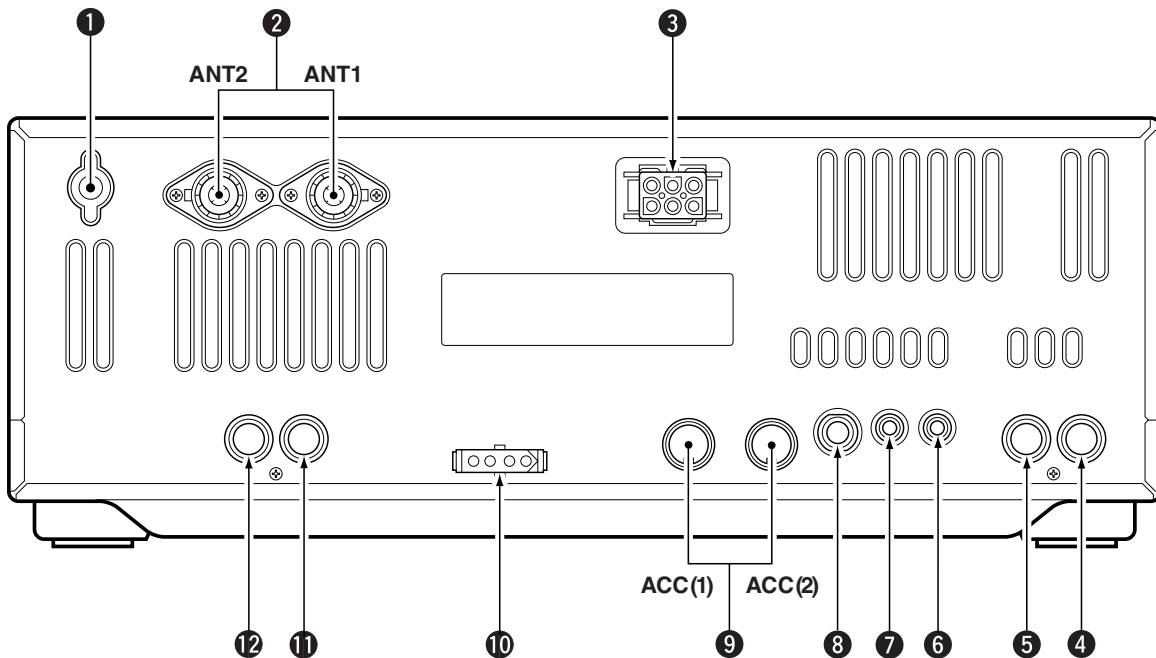
- 39 RECEIVE INDICATOR [RX]
- 40 TRANSMIT INDICATOR [TX]
- 41 REC/PLAY SWITCH [REC/PLAY]
- 42 LOCK/SPEECH SWITCH [LOCK/SPEECH]
- 43 TUNING DIAL
- 44 LOCK INDICATOR [LOCK]



**G**

- 45** PASSBAND TUNING CONTROLS [TWIN PBT]
- 46** PBT CLEAR SWITCH [PBT CLR]
- 47** NOTCH SWITCH [NOTCH]
- 48** MANUAL NOTCH FILTER CONTROL [NOTCH]
- 49** CW PITCH CONTROL [CW PITCH]
- 50** ΔTX SWITCH [ΔTX]
- 51** RIT SWITCH [RIT]
- 52** RIT/ΔTX CONTROL [RIT/ΔTX]
- 53** CLEAR SWITCH [CLEAR]

### 3-2 Rear panel



- 1** GROUND TERMINAL
- 2** ANTENNA CONNECTOR 1 [ANT 1]/  
ANTENNA CONNECTOR 2 [ANT 2]
- 3** DC POWER SOCKET [DC 13.8V]
- 4** SEND CONTROL JACK [SEND]
- 5** ALC INPUT JACK [ALC]
- 6** EXTERNAL SPEAKER JACK [EXT SP]
- 7** CI-V REMOTE CONTROL JACK [REMOTE]
- 8** STRAIGHT KEY JACK [KEY]
- 9** ACCESSORY SOCKET 1 [ACC (1)]/  
ACCESSORY SOCKET 2 [ACC (2)]
- 10** TUNER CONTROL SOCKET [TUNER]
- 11** RECEIVE ANTENNA CONNECTOR [RX ANT]
- 12** TRANSVERTER JACK [XVERT]

## 4. What is DSP in radio communication?

The term DSP stands for “digital signal processor”. When DSP is used in a communication unit, the electrical signal processes (amplification, filtering mixer, modulation, demodulation, etc.) are handled by the DSP. Such signal processing, using numerical calculations, is called “digital signal processing”.

Digital signal processing assures the same results every time providing for the characteristics defined in the design phase. When digital signal processing is utilized, it is not necessary to take the adjustment deviations of the conventional analog circuit into consideration. These deviations are caused by variations in component characteristics, temperature change, or deterioration over time. It is also possible to perform complicated processing tasks such as Fourier transformation, adaptive control, special function processing (\*1), and more. Such complicated processing tasks are very difficult and costly for a conventional analog circuit.

\*1 Special function processing: Trigonometric function, inverse function of trigonometric function, square root, logarithmic function, exponential function, etc.

Digital signal processing is also widely used in fields other than radio communication units, such as:

- Modems for telephone circuits
- Surround-sound effects (stereo sets, stereo components)
- Echo canceller (telephone)
- Voice compression/coding (cellular phones)

It is possible for a computer CPU to execute digital signal processing. However, a DSP differs from a CPU in that it has the dedicated hardware construction required for the effective execution of digital signal processing. Basically the unit has a multiplication/addition circuit widely used for DSP to execute the combination of multiplications and additions in one clock, and with an internal data bus of more than two circuits, to fetch two data items required for calculation at the same time. It also has a loop processing function to execute repeated calculations with high efficiency and a data address creation function to transmit signal data effectively, which are assigned to consecutive addresses. These functions are incorporated as dedicated hardware.

Because their performance has developed quickly, the CPUs currently used for computers can execute digital signal processing. A CPU with a high clock frequency may be superior in calculation performance to a low-end DSP chip. When it is compared with a CPU of equivalent performance, a DSP with hardware specialized to digital signal processing has the following advantages:

- Low clock frequency
- Low integrity (reduced logic scale)
- Low power consumption (Low heat energy generated)
- Low cost

When various judgment functions are required, or when different calculations are repeated each time, a DSP is not suitable. The CPU is then more suitable for such processing.



DSP chip

### Background to development of the first-generation IC-756PRO

From the early stages of research into DSP transceivers Icom has been on the forefront of shifting IF filter design from analog to digital type filters. To put the digital IF filter to practical use it was necessary to incorporate the digital IF filter into the AGC loop. It was also necessary to provide AGC processing using the DSP. To achieve this there remained a lot of technical problems to be resolved.

In the initial stage of research, it was not possible to complete the DSP, A/D, and D/A devices in a radio unit at a practical cost, as shortage in device capacity was a significant factor. Icom conducted research into digital PSN modulation, noise reduction, automatic notch, and audio peak filter, while also proceeding with research into digital IF filter processing and digital AGC processing. This research includes the ultra-narrow filter for CW that allows the advantages of DSP to be fully utilized for commercialization of a DSP transceiver. As the first devices were developed with improved capacity, Icom started full-scale research into integrating the digital IF filter processing (\*2) and digital AGC processing in practical applications.

\*2 Advantages of digital IF filter processing:

Since a digital IF filter is free from deterioration due to passband width deviations, temperature changes, change in mechanical strength, etc., the changes seen in an analog filter will not occur. It will not deteriorate through years of use and will provide excellent ripple characteristics that are not possible with analog filters.

When the DSP filter is processed at the AF stage, the demodulated AF signal is filtered after this. This filter type will function effectively when the level of the interfering signal is equal to or less than that of the desired signal. However, when the level of the interfering signal increases, the AGC activates reducing the level of the desired signal causing it not to be heard (AGC blocking phenomenon). This phenomenon is caused by filtering taking place outside the AGC loop. Even if filtering is executed at the IF stage before demodulation, it is not possible to avoid this blocking phenomenon when the digital filter is not incorporated in the AGC loop. Therefore, it is necessary to execute both IF filter processing and AGC processing using the DSP to prevent the AGC blocking phenomenon.

To realize a digital AGC, it is necessary to obtain the adjustment range for AGC gain internally in the DSP (\*3), and to input both the desired signal and the interference signal into the A/D converter without them distorting (\*4). For these reasons, Icom decided on a dynamic range for the A/D converter of at least 110dB, and approximately 120dB when the margin is taken into consideration.

\*3 To control the AGC attack response properly, it is necessary to adjust the gain even after the completion of IF filter processing. If the adjustment range of gain within the DSP is set to 60dB, it is necessary to obtain a wider dynamic range, as the noise floor is raised 60dB under full-gain conditions where AGC is not applied.

\*4 If the signal is distorted before entering the A/D converter, a distortion component may be mixed in the band. If it is mixed in the band, it is extremely difficult to remove it by post processing.

The DSP in the IC-756PRO/756PROII employs a 24-bit A/D converter. The logical value of the dynamic range of a 24-bit A/D converter is 144dB, however the actual value of the analog performance is smaller than this and performance may differ considerably, depending on the type of A/D converter used.

The A/D converter used for the IC-756PROII is a super-high performance A/D converter that is also used in digital mixers for recording studios and provides an actual analog performance value of 120dB. To bring this performance to an optimum level it is necessary to execute calibration for 10 seconds after powering on. The wait time when IC-756PROII is started is allotted to the calibration operation.

To execute the processing of data sampled by the 24-bit A/D converter it is necessary to obtain 24-bit calculation accuracy. Since the dynamic range is decreased substantially due to the scaling operation (\*) for the accumulation of calculation errors or digital filter processing, Icom felt the 24-bit fixed decimal point DSP would provide insufficient calculation accuracy.

\* Scaling

For digital filter processing, a frequency which causes the gain to increase may exist at the intermediate stage of processing even if the filter used provides a passband gain of 0dB. For a fixed decimal point, DSP the calculation is executed with the gain decreased in advance so as not to allow an overflow to occur due to a signal of that frequency. This gain adjustment operation is called "scaling".

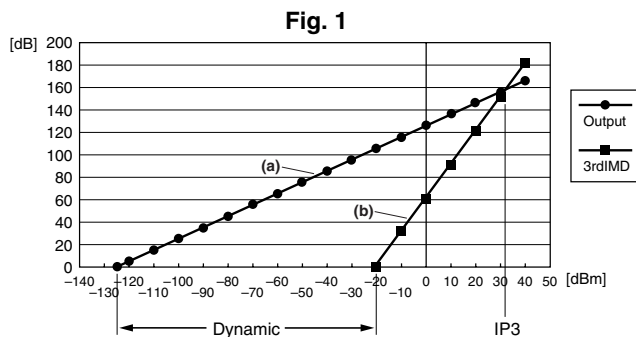
Since the level of scaling required is also increased to provide an IF filter with a sharp shape factor the calculation accuracy is liable to be decreased, even if double-precision (32-bit fixed decimal point) calculation is executed when using a high-speed 16-bit DSP. To provide both the digital IF filter processing and digital AGC processing using DSP, Icom determined it was necessary to use a 32-bit floating point DSP.

For a 32-bit floating point DSP, the numerical data within the DSP is adjusted automatically according to the size of the numerical value. Consequently, errors generated due to calculation are extremely limited and the influence of calculation errors is almost negligible. Because it is not necessary to consider the overflow during calculation, the dynamic range will not be decreased due to the scaling operation.

The 32-bit floating point DSP and 24-bit A/D-D/A converter use a signal processing algorithm (newly developed to demonstrate its performance) in combination for the reasons above, which make it possible to provide highly accurate digital IF filter processing and digital AGC processing. These new functions (FM demodulation, AM modulation/demodulation, RTTY modulator, etc.) were incorporated in the IC-756PRO to make it an IF DSP radio.

## Two Dynamic ranges

### Dynamic range as RF performance



"Dynamic range" from the viewpoint of RF performance indicates to what extent the distortion component (generated due to the frequency of a signal) can be heard at the receive frequency when a frequency component different from two receive frequencies is input. "Dynamic range" generally means the value by 3rd order distortion component.

If the receive frequency is substituted for "f<sub>RX</sub>", the input frequency for "f<sub>1</sub>" and the input frequency for "f<sub>2</sub>" respectively, the following relationship is established for 3rd order distortion component.

$$f_1 \times 2 \pm f_2 = f_{RX}, \text{ or}$$

$$f_1 \pm f_2 \times 2 = f_{RX}$$

If there are inputs of 14.2MHz and 14.3MHz while 14.1MHz is being received, the distortion component is heard at 14.1MHz. The relative value of the input level when the signal can be heard at 14.1MHz and the level of the signal received at the essential receive frequency is called the "dynamic range".

Figure 1 shows an example in which the following are plotted on the same axis.

- Input/output characteristics at receive frequency, or the characteristic data (a) for a case when the receive frequency component input from the ANT is detected and output as a low frequency signal
  - Input level of frequency component (generating 3rd order distortion from the receive frequency) and level (b) at which the distortion can be heard at the receive frequency.
- The difference in level at which (a) and (b) above can be heard is the dynamic range.  
The level at intersecting point between (a) and (b) above is called IP3 (3rd order intercept point).

If these numerical values are large, it can be said that signal processing is executed without distortion. When the numerical values are small, a frequency component that does not exist in the essential receive frequency is heard and distortion will be generated.

### Dynamic range for A/D converter

Consider the dynamic range (used as an index for the performance of an A/D converter) as the ratio between maximum value and minimum value to be treated by the A/D converter. If the maximum resolution for one bit is "V<sub>max</sub>" in the case of a 16-bit A/D converter, the following is given:

$$V_{max} \div 2^{16} = V_{max} \div 65536$$

In other words, the change in level for one bit is 1/65536 of V<sub>max</sub>. This value seems to be an extremely small value, in decibels it will be as follows:

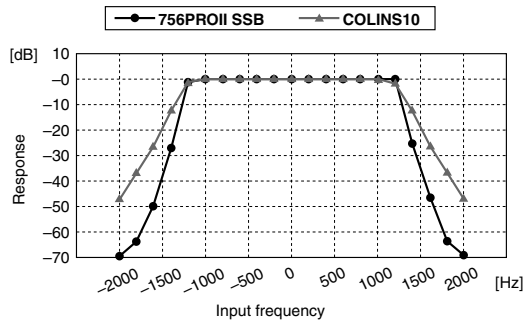
$$20 \log (1/65536) = -96.33 \text{dB}$$

This means that an S/N ratio of over 96dB is never allowed for transmit. The minimum resolution of signal the A/D converter can treat is affected by its specifications, which are 24 bits and 144dB logical value. Some may say that a transceiver is not a high-grade audio system and therefore does not require a specification of 144dB, or that a specification of 96dB is sufficient; however this value is not an over specification. If there is no AGC in the DSP and the input level of the A/D converter is properly controlled by the analog circuit AGC, the specification of 96dB will be sufficient (the IC-775DSP uses this system). When the A/D converter is in the AGC loop, the input level of the A/D converter may fluctuate significantly. For this reason, the gain control by AGC within DSP requires at least the dynamic range of the A/D converter.

## 5. Circuit description

### 5-1 Digital IF filter

For IC-756PROII the transmit/receive passband width in all modes is determined by the digital IF filter using DSP. A filter of this type provides an ideal shape factor that cannot be achieved by an analog filter. If an attempt is made to increase the shape factor and band ripple characteristics of an analog filter, it is necessary to increase the number of crystal components (or ceramic elements), which may result in physical restrictions. A digital IF filter using DSP assures the desired characteristics by overlaying multiple filters. This is governed only by the processing volume of the software and it is possible to overlay such filters with any number of stages.



The diagram shows a graph of receive selectivity when the IC-756PROII is set to the SSB BW mode of 2.4kHz as well as the selectivity characteristic of each Collins 10-pole mechanical filter. The digital IF filter of the IC-756PROII is of a design equivalent to a 14-pole filter. The filter serves to cut the undesired adjacent signals sharply under any circumstances using the superior shape factor (sharp/soft) and 51 types of variable passband width provided by IF stage processing using the DSP. When viewing a received CW signal the difference between the cut-off performance of this filter and that of an analog IF filter is evident.

In a transceiver equipped with a conventional analog IF filter the beat frequency of an adjacent signal is present when the CW signal is received resulting in interference. The beat frequency is contained in the skirt of the filter even if it is out of the set band range. (Fig. 1.1)

Fig. 1.1

#### Conventional analog filter

The signal out of setting band is also heard as CW reception sound.

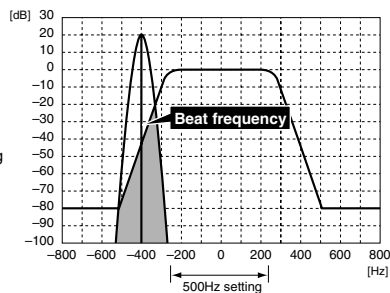
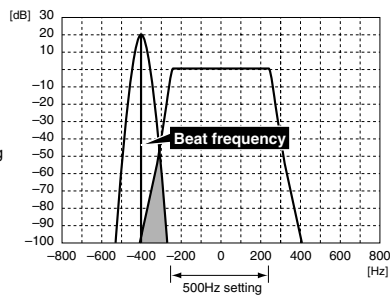


Fig. 1.2

#### Digital filter of IC-756PROII

The signal out of setting band is not heard as CW reception sound.

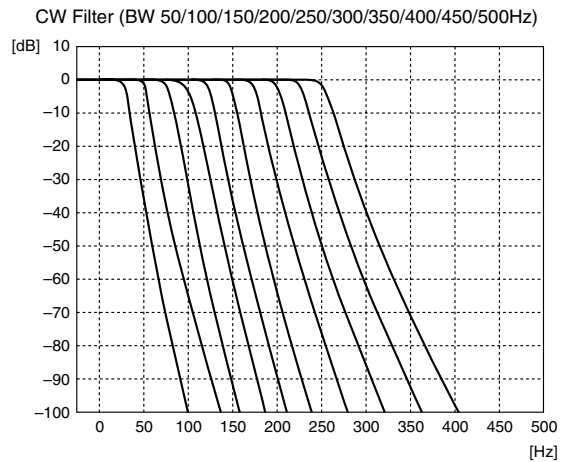


When using a digital IF filter the beat frequency of an unwanted adjacent signal moves out of the filter passband width, which will not cause interference. (Fig. 1.2) This is the greatest difference between an analog IF filter and a digital IF filter. During "pile-ups", such as those that occur in DX'peditions, contests, etc., it is possible to make a proper selection suitable to the application by selecting the broad filter shape (SOFT).

#### 5-1-1 CW sharp filter

The digital IF filter offers an ideal shape factor which has never been available with conventional analog filters. It enables a greater ability to receive weak stations that may lie behind radio interference. This is the filter shape that Icom would suggest to the DX hunter due to its superior cut-off performance. The cut-off performance is of a level to actually extend the CW band as explained above.

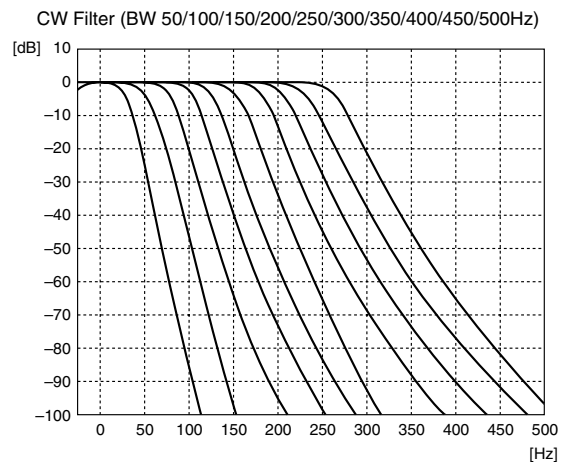
#### CW sharp filter characteristic



#### 5-1-2 CW soft filter

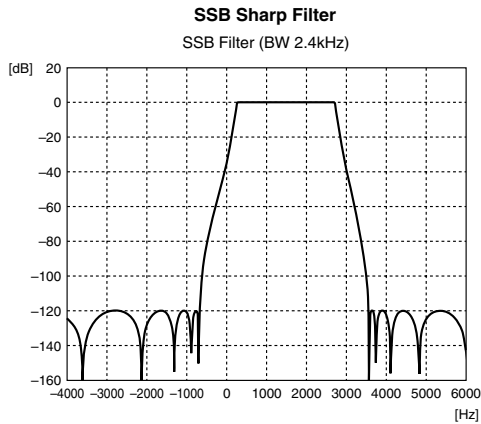
The skirt characteristics of the soft filter are broadened so that the listening level of the filtered signal is the same level as that of a conventional analog filter. When using the radio for DX'pedition the filter is recommended for "pile-up" operation and is most suitable for the CW DX'peditioner and CW contestant.

#### CW soft filter characteristics



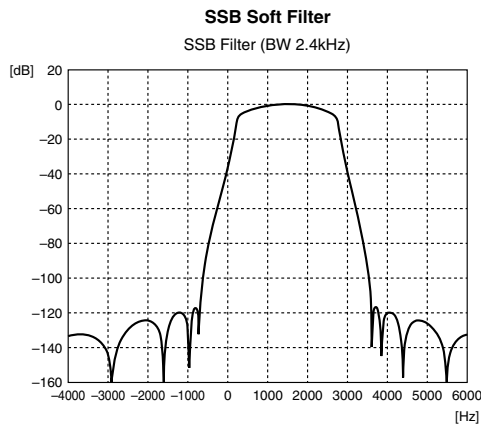
### 5-1-3 SSB sharp filter

This filter creates an ideal shape factor and in-band flatness, and makes it possible to cut out-of-band signals while reproducing the in-band signal, without deteriorating sound quality. This filter shape is most suited for situations which emphasize ragchewing and receive sound quality.

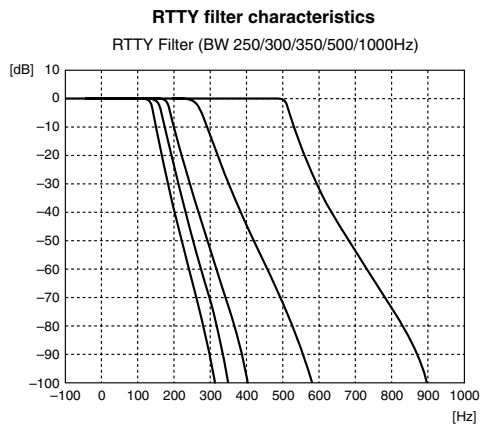


### 5-1-4 SSB soft filter

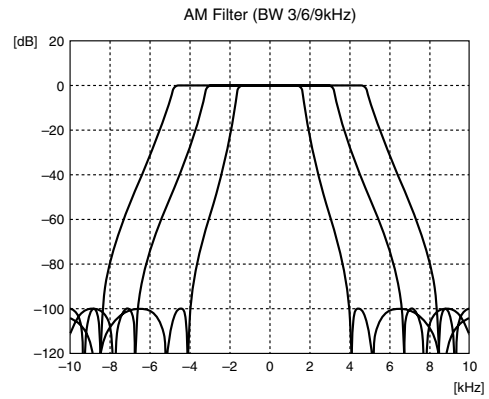
The soft filter shoulder is rounded to provide a receive sound approximating an analog filter. The noise is reduced for high-pass and low-pass to improve the S/N ratio for the desired signal. This function will demonstrate its effect when the signal closest to the noise level is picked up in the 50MHz band. Since the desired skirt characteristics are maintained it assures superior filtering performance.



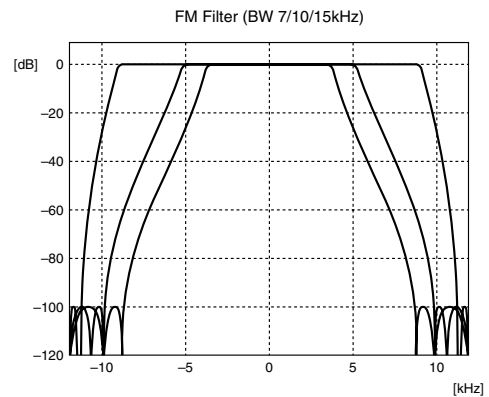
### 5-1-5 Other digital filters



### AM Filter Characteristic



### FM Filter Characteristic



### Digital IF filter transmission band (51 types)

Application mode	FILTER	Standard values	Setting range (step width)
SSB	FIL1	3.0kHz	50-500Hz (50Hz) / 600-3.6kHz (100Hz)
	FIL2	2.4kHz	
	FIL3	1.8kHz	
SSB*D CW	FIL1	1.2kHz	50-500Hz (50Hz) / 600-3.6kHz (100Hz)
	FIL2	500Hz	
	FIL3	250Hz	
RTTY	FIL1	2.4kHz	50-500Hz (50Hz) / 600-2.7kHz (100Hz)
	FIL2	500Hz	
	FIL3	250Hz	
AM	FIL1	9.0kHz	—
	FIL2	6.0kHz	
	FIL3	3.0kHz	
FM	FIL1	15kHz	—
	FIL2	10kHz	
	FIL3	7.0kHz	

## 5-2 Digital functions

### 5-2-1 Noise reduction, automatic notch

An adaptive filter made up of an FIR filter and LMS algorithm as shown in Fig. 2 is used to provide the basic configurations of noise reduction and automatic notch. This adaptive filter (\*) separates the target signal and noise, the correlation of separation parameters, and controls the coefficient of the FIR filter with the LMS adaptive algorithm to minimize the error between the output of the FIR filter and the reference signal.

\* Adaptive filter

This type of filter is called an "adaptive filter" since the filter characteristics are changed by adapting to the characteristics of the input signal.

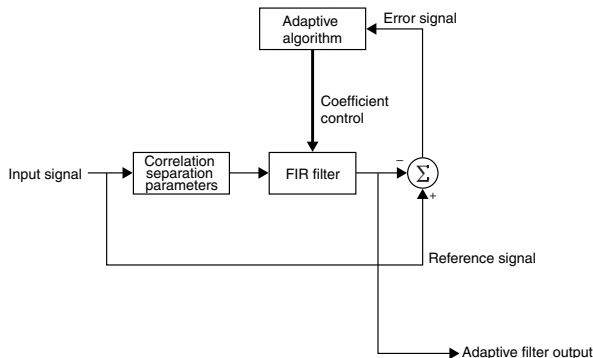


Fig. 2

### 5-2-2 Noise reduction

The adaptive filter allows the target signal to pass while the noise component (random signal) is attenuated. The voice signal has a high short-time correlation and a low long-time correlation. (For discrimination the signal correlation is called "short-time" or "long-time" for convenience, however it is a difference of only several hundreds microseconds.) If the correlation separation parameters are set to allow short-time correlation to be detectable, the voice signal is detected as a low correlation component, and the noise as a high correlation component. In this case the voice component must pass as it is, and only the noise component is attenuated. The noise reduction effect is random at the head of a word (the moment when speech begins) and at points where intonation changes significantly. If the noise reduction effect is increased too much, the voice component may be attenuated together with the noise as described above. In this case it may decrease clarity, even if the S/N ratio is improved.

The transceiver is designed with the flexibility to set the noise reduction level accurately (16 stages) in order to meet all circumstances. This makes it possible to adjust the balance between the S/N ratio and clarity quickly.

### 5-2-3 Automatic notch

If the correlation separation parameters are so set to allow long-time correlation to be detectable, the voice signal is detected as a low correlation component, and only the tone signal is detected as a high correlation component. If the correlation separation parameters are set to allow long-time correlation to be detectable the voice signal is detected as a low correlation component, and only the tone signal is detected as a high correlation component. Since this setting makes it possible to separate the tone signal component from the voice component the output from the adaptive filter will be only the tone signal. Since the phase and amplitude of the tone signal from this adaptive filter become the same as those of the input signal, the output of the error signal shown in Fig. 2 makes it possible to obtain a voice signal from which the tone signal is removed. In other words, this adaptive filter setting will operate as an automatic notch to remove beat interference such as CW and RTTY signals, which may interfere with SSB. Automatic notch makes it possible to detect and

remove interference correctly even when more than two tones occur. As the tone frequency changes the interference is followed and removed automatically. Since the characteristics are adjusted to minimize the influence upon a voice, it can be used in SSB mode without any sense of incongruity even if automatic notch is turned on all the time.

## 5-3 PSN modulation

The IC-756PROII adopts a digital PSN modulation system for SSB modulation processing to provide superior band characteristics and a high transmission S/N ratio. This means that unwanted sidebands and carrier leaks are almost completely eliminated. This section explains the principle of operation while comparing the PSN type SSB modulator with the analog filter type SSB modulator used in conventional analog transceivers.

### 5-3-1 Analog filter type SSB demodulator

The configuration of analog filter type SSB modulator is shown in Fig. 3.

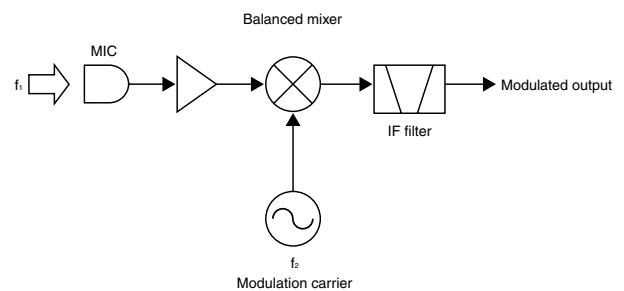


Fig. 3

If the tone signal of frequency ( $f_1$ ) is presented to a microphone, two spectra ( $f_2-f_1$  and  $f_2+f_1$ ) are generated against the mixer output as shown in Fig. 4.

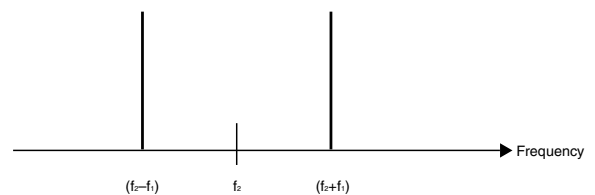


Fig. 4 Mixer output spectrum

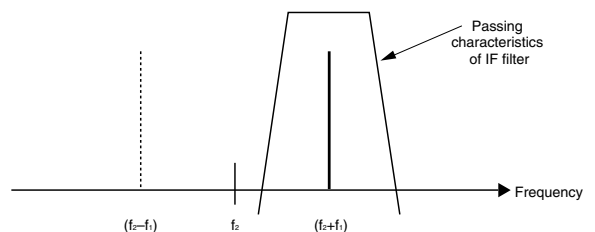


Fig. 5 SSB demodulated wave

This mixer output passes through the IF filter, passing only the necessary band. Its unwanted sideband is attenuated, which assures a modulated SSB signal. (Fig. 5)

Since the performance limits of the IF filter become the performance limits of the modulator in an analog filter type SSB modulator (Fig. 3 shown on page 14) the problems below will exist:

1. The ripple characteristics within the passband of the IF filter is reflected directly upon the entire frequency characteristic of the modulator.
2. There is a limitation in the shape factor of an IF filter. If an attempt is made to execute the modulation output to be excessively low-bandwidth it becomes unable to fully restrict the unwanted sideband signal.
3. A crystal IF filter with a good shape factor may not provide the satisfactory group delay characteristics in many cases and may be inferior from the viewpoint of sound quality.

### 5-3-2 PSN type SSB modulator (basic type)

The PSN type SSB modulator uses phase shift operation to negate the unwanted sideband signal and to attain a modulated SSB signal. If it is possible to reduce the phase difference at low-band of a 90° phase shifter it will assure superior characteristics to the filter type SSB modulator as it is possible to attain a higher unwanted sideband signal suppression ratio compared with that at low-band.

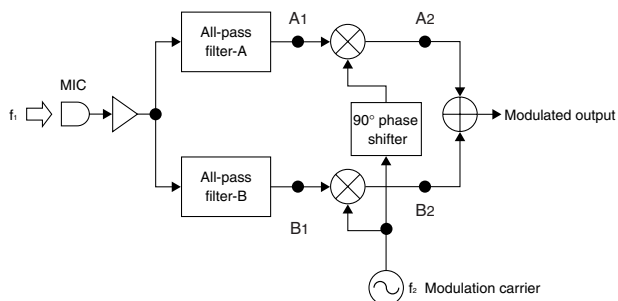


Fig. 6 Configuration of PSN type SSB modulator

The two filters (filter A and filter B) shown in Fig. 6 are combined to make a 90° phase shifter. This is an all-pass filter (\*) designed using two filters in pairs so that the signal output from each filter appears to have a phase difference of exactly 90° when the same signal is input.

\* All-pass filter:

An all-pass filter is used to change only the phase without changing the amplitude of the signal sent from the all-pass filter.

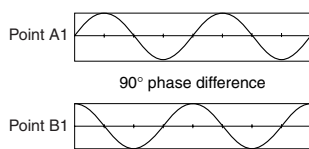


Fig. 7 Output signal of all-pass filter

When a signal (frequency:  $f_2$ ) having a phase difference of 90° against the signal output from the all-pass filter of two lines (A, B) transmitted from a station is modulated with the tone signal of frequency ( $f_1$ ) presented to the microphone, two spectra ( $f_2-f_1$  and  $f_2+f_1$ ) are generated at two points, point A2 and point B2 respectively, each of whose phase relationship is as shown in Fig. 8. The signal at point A2 is added to that at point B2 as indicated. The sideband signals having a phase difference of 180° are negated while the sideband signals of the same phase add with each other, causing an output whose amplitude is doubled. The example shown in Fig. 8 shows a USB signal that is obtained.

When an LSB signal is required for modulated output, it is best to add it after inverting the polarity. Since the component with a 180° phase difference is replaced with that having the same phase the modulated output appears to be an LSB signal.

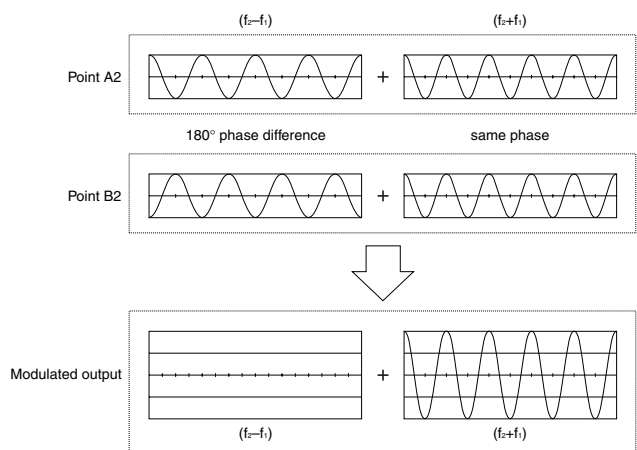


Fig. 8 Phase relationship of modulated signal

The PSN type SSB modulator provides an SSB modulated signal by eliminating the unwanted side band component. To achieve this it is necessary to keep the phase difference accurately and to set the amplitudes to precisely the same level.

With a PSN type SSB modulator using analog circuit, such problems as changes in characteristics due to deviation in parts or temperature may occur. Accordingly, it is very difficult to achieve the same unwanted sideband signal restriction ratio with a filter type SSB modulator.

For these reasons few transceivers adopt the analog type PSN.

Using the DSP it is possible to provide stabilized performance even if the PSN method is used, as it has few of the fluctuations seen in the analog circuit.

### 5-3-3 Icom's PSN type SSB modulator

Figure 6 is a basic configuration drawing of a PSN type SSB modulator. The IC-756PROII adopts the PSN type SSB modulator using Icom's unique architecture shown in Fig. 9.

This method makes it possible to obtain an effect equivalent to multi-rate processing (\*) even if the sampling rate is not decreased during all-pass filter processing. This makes it possible to improve the DSP calculation by more than two times which is required for accurate SSB modulation processing. The part of modulated carrier multiplication in the conventional method is changed to the multiplication of a constant leading to an improvement in efficiency.

\* Multi-rate processing

A method of processing that uses the multiple sampling rates selectively, depending on the frequency of signal to be processed.

Even if the processing contents are the same, the processing of a lower sampling rate will decrease the volume of calculations.

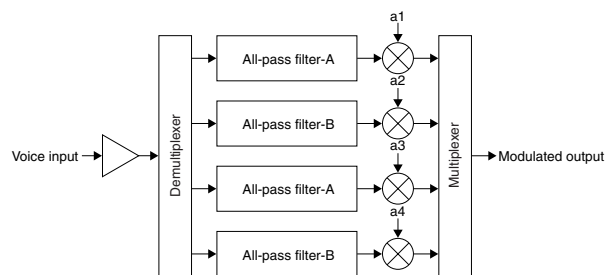


Fig. 9 Configuration of Icom's PSN type SSB modulator

For the SSB modulator shown in Fig. 9, the signal is input to each filter sequentially for each sampling cycle by using a multiplexer with 4 all-pass filters (filter A, filter B, each designed for a phase difference of 90°) arranged alternately to multiply each filter output by the constants (a1 to a4).

Using the multiplexer the result of multiplication is output sequentially, making it possible to gain the desired SSB-modulated output signal.

- For USB: Constant {a1, a2, a3, a4} = {1, 1, -1, -1}
- For LSB: Constant {a1, a2, a3, a4} = {1, -1, -1, 1}

For PSN modulation processing using the 16-bit fixed decimal point DSP of conventional transceivers, the characteristics are adjusted to decrease the influence of the rounding error (when the filter coefficient is quantized) as it occurs. For the 32-bit floating point DSP nearly ideal characteristics are assured as the influence of errors due to quantizing is extremely limited. The IC-756PROII was re-designed with this point taken into consideration to further improve the low-band characteristics as compared with conventional transceivers. Figure 10 shows the restriction characteristics of an unwanted sideband signal and the pass characteristics of the desired sideband signal.

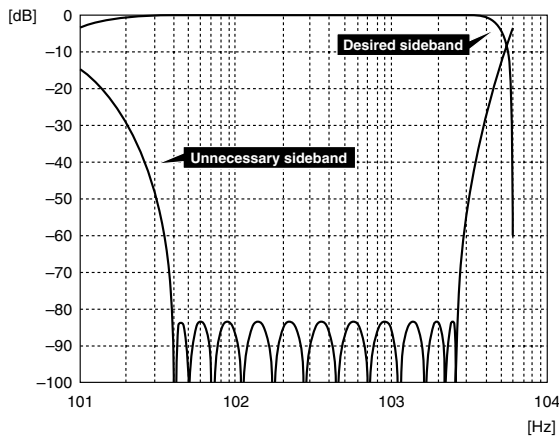


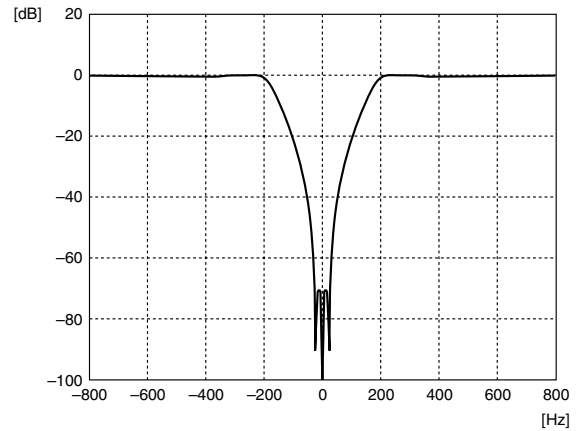
Fig. 10 SSB modulation characteristics

#### 5-4 Manual notch

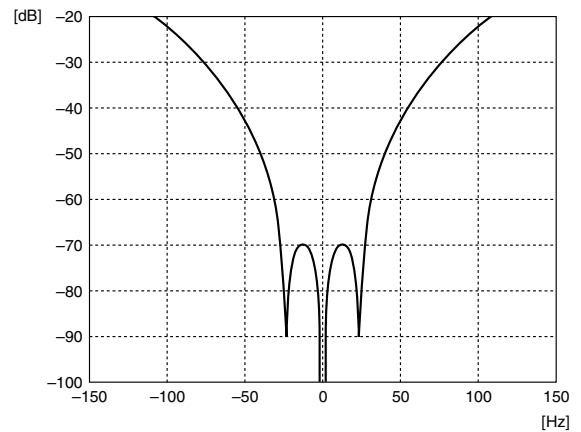
The IC-756PROII manual notch filter has extremely sharp characteristics which can be provided only by DSP processing. Since this manual notch is processed within the AGC loop even powerful beats are cut-off sharply without any influence upon the AGC. The filter characteristics are sharp and the passband width is held to approximately 50Hz with an attenuation level of over 70dB. This makes it possible to adjust the notch point accurately. Only the DSP provides the characteristics as shown above.

With an analog type notch filter (crystal or LC notch filter) it is not possible to adjust the notch point characteristics accurately as shown above, as the frequency characteristics are liable to deviate. The manual notch assures stable filter characteristics by DSP processing because of its extremely sharp characteristics and the high-stability reference oscillator provides superior frequency stability.

Accordingly it provides stable operation such that it is not necessary to re-adjust the notch point, provided the beat signal is not moved once it is set.



Characteristics of manual notch



Characteristics of manual notch (enlarged view)

#### 5-5 Speech compressor

The IC-756PROII is equipped with a newly developed RF type speech compressor. The configuration of the speech compressor is shown in Fig. 11.

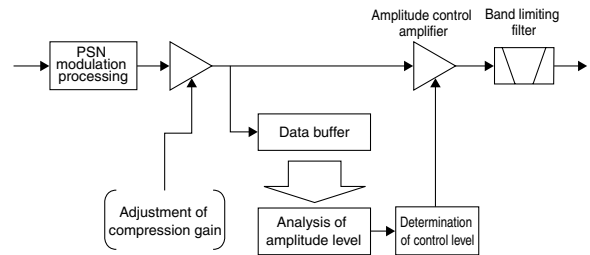


Fig. 11 Configuration of speech compressor

The operating principle of this compressor is that the SSB-modulated IF signal is saved in the data buffer for a fixed time at first, and then the IF signal saved in the buffer is analyzed for amplitude level. The control level of the amplitude control amplifier is determined in accordance with the analysis, providing compression control such that the signal peak does not exceed a certain level. In other words, the amplitude of the current signal is controlled in accordance with the change in amplitude over a certain previous period.



Unlike the RF compressor used widely in conventional analog processing type transceivers little distortion will occur as the signal is not clipped. The speech compressor resembles an AGC type compressor in that the signal level is controlled, however the normal AGC method has a lot of problems. It is usually considered that the AGC type has an improved compression effect along with shortened gain recovery time constant, compared to the grip type. Setting the time constant to a low level may bring about an inferior compression effect as the adjustment range of the time constant is limited due to spoiled AGC loop stability.

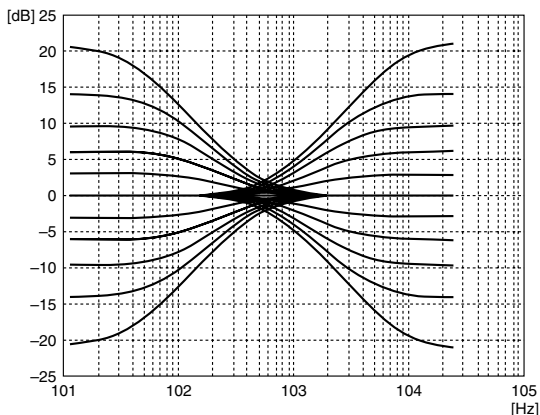
The Icom type compressor assures a high compression effect as there are no problems due to the non-execution of feedback processing with a proper follow-up performance against changes in amplitude of the IF signal. Even when the compression level is high only a slight distortion outside audible range may occur. To prevent the transmit passband width from extending a wide-band limiting filter is used. Since this filter was designed to prevent group delay degradation, it does not have an influence upon the modulated sound quality.

### Distortion generated by compressor processing

For distortion generated by compressor processing, only the high order distortion may be addressed in many cases. Also, mutual modulation distortion may occur when the input signal is of 2 tones or more. The RF stage grip-processing compressor is better than the AF stage grip-processing compressor from the viewpoint of high-order distortion. The reason why it is not so highly rated from the sound quality viewpoint is because there is a problem with mutual modulation distortion. The AGC type compressor provides a lower mutual modulation distortion level as compared to the grip-processing compressor assuring better sound quality. The Icom type restricts mutual modulation distortion similarly.

## 5-6 Microphone equalizer

The microphone equalizer characteristics used for the IC-756PROII are based on the frequency characteristics of the audio tone control circuit which has been re-designed to be dedicated to voice frequency range. The transmit function of an analog filter is simulated and converted into that of a digital filter to provide the microphone equalizer function. In some microphone equalizers for transceivers the characteristics may change suddenly with a specific frequency as a limit. Unnatural sounds may be generated by such equalizers depending on their tone quality. Not in Icom's.



Characteristics of microphone equalizer

The microphone equalizer of IC-756PROII allows smooth selection of characteristics and may be adjusted accurately over 11 stages for high band and low band. This makes the frequency characteristics adjustable without any sense of incongruity.

## 5-7 RTTY demodulator

The IC-756PROII is equipped with a built-in demodulator/decoder function (for BAUDOT RTTY) for the first time in an HF amateur transceiver. It is possible to decode RTTY signals using the transceiver independently even if external units such as multi-function TNC, and a RTTY terminal unit (compatible to RTTY) are not used. When the RTTY signal is decoded the DSP unit executes the demodulator processing and the binary signal (BAUDOT) obtained is decoded by the main CPU, and its characters are displayed in the lower portion of the display. Figure 12 shows the configuration of demodulator processing using DSP.

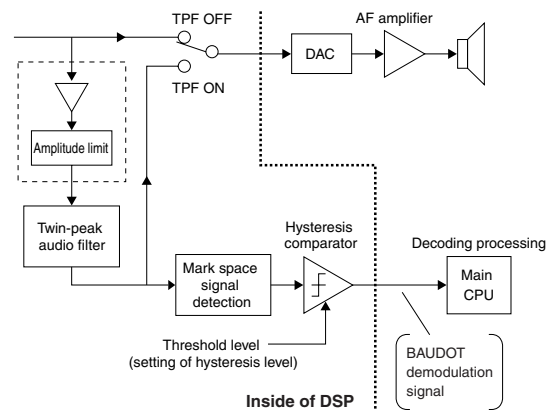


Fig. 12 Configuration of RTTY demodulator

Most conventional RTTY terminal units or TNCs use either the PLL type or filter type demodulator to detect the mark/space signal. When the communication conditions are undesirable due to interference, fading, etc., the filter type is generally superior. The demodulator processing of IC-756PROII uses the basic configuration of a filter type demodulator.

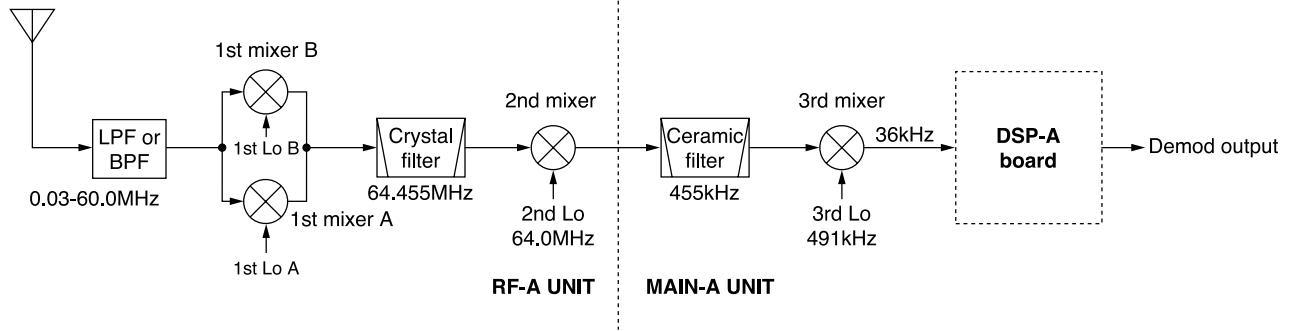
For demodulator processing in DSP the amplification and amplitude limitations are first executed against the audio signal demodulated through product detection. This processing provides sufficient demodulation performance against even low level signals that do not move the S-meter, so there is no influence due to deviations in amplitude. The twin-peak audio filter then removes the radio interference and improves the S/N ratio before detecting mark/space signals. Two narrow-band pass filters are used in detection processing to extract the components near the mark frequency and space frequency. The output of each filter is detected and balanced, polarity reversed, and then passed to comparator processing. The comparator processing has a hysteresis characteristic such that it is hardly affected by fluctuations in the noise component. The hysteresis width is adjustable by changing the threshold level value on the RTTY decoding screen. The comparator determines the signal for polarity. The result is converted into a logic signal and then transmitted to the main CPU. The main CPU decodes the RTTY signal and displays the characters on the display screen.

In filter type demodulators, the difference in filter characteristics appears to be a difference in decoding performance. The filter will enhance the decoding ratio provided a high performance filter is used. It is also influenced by the phase and time response characteristics. Twin-peak audio filters and mark/space signal detection filters are carefully tested to adjust their characteristics. Final development of the Icom filter was conducted in part in cooperation with veteran stations with a long RTTY history. A decoding ratio equivalent to a dedicated RTTY unit such as TNC or RTTY terminal units designed for existing RTTY is achieved. For the

IC-756PROII RTTY demodulator the effect of twin-peak audio filtering has made a significant contribution to improving the decoding ratio. When the RTTY mode setting is selected, it becomes possible to change the speaker output and the audio output through the accessory terminal to a signal filtered by the twin-peak audio filter. Using this function it is possible to improve the decoding ratio of a TNC, terminal unit, etc. connected to the radio. Since the twin-peak audio filter is connected at all times to the built-in demodulator, it is not necessary to set the twin-peak audio filter output when using only the built-in demodulator.

## 5-8 Receiver

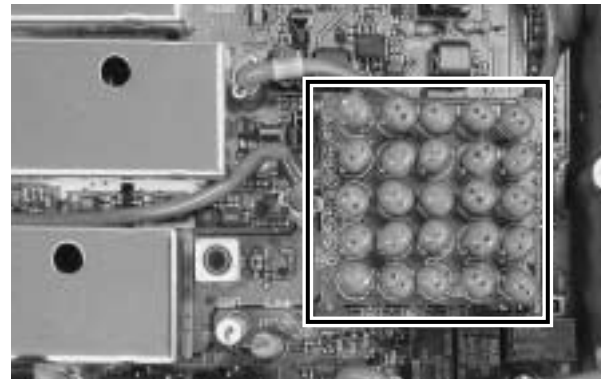
### Receiver block diagram



The signal received at the antenna terminal (ANT1/ANT2) passes through the antenna selector and enters the RF-A unit through the LPF built into the CTRL unit. When the antenna tuner is turned on the IC-756PROII removes interference and cross modulation from unwanted radio signals to some degree in the first stage during receive, using the coil/capacitor of antenna tuner, and by allowing the signal to pass through the matching circuit. The signal input to the RF-A unit passes through the relay selectable ATT circuit (6/12/18dB) and is lead to the BPF stage which is divided into 13 sections. Various frequency components are included in the input side of the BPF stage. When distortion occurs in the BPF stage input side the distortion component may enter the band resulting in an interfering signal. However high-performance BPF may be used. The PIN diode with wide-range frequency characteristics and limited secondary distortion (Motorola, MMBV3700) is used to restrict such distortion. In addition, a large-sized coil (L) is used in the BPF stage element. The capacitor (C) provides low conductivity and low distortion. This prevents the IMD characteristic from being deteriorated by the filter and significantly improves the performance against the influence of adjacent intensive electric fields and weak signals.

#### 13-division BPF stage

Band	Control signal	Band	Control signal
0.03-1.6MHz	B0	11-15MHz	B7
1.6-2MHz	B1	15-22MHz	B8
2-3MHz	B2	22-30MHz	B9
3-4MHz	B3	30-50MHz	B10W
4-6MHz	B4	50-54MHz	B10
6-8MHz	B5	54-60MHz	B10W
8-11MHz	B6		



BPF stage

Having passed through the BPF stage the signal enters the preamplifiers (2 types). Preamplifier 1 is a GG (granted gate) amplifier of push-pull configuration instead of the conventional FET gate-earth type parallel amplifier. Preamplifier 2 is designed with gain for high-bands emphasized and is suitable for antennas with increased loss, small-loop type antennas having a limited band, and compact type YAGI antennas. The gain is set to approximately 10dB for preamplifier 1 and to approximately 16dB for preamplifier 2.

After passing through a preamplifier, the signal enters the parallel GG (granted gate) amplifier arranged at the front of the 1st mixer. This amplifier compensates for the loss of the splitter circuit for dual watch and isolates the main mixer from the sub-mixer. This signal enters the 1st mixer through the GG amplifier.

The mixer circuit incorporates a double balanced mixer in which four FETs are used to provide high IP and high dynamic range. This provides a significant improvement of the S/N ratio with limited distortion against large input signals, and provides superior 2-signal characteristics with no influence from the strong signals of an adjacent frequency.

The 1st mixer and LO circuits are arranged in two sets to provide the dual-watch function. The signal is converted to 64.455MHz by the 1st mixer and then passes through a variable type attenuator (using the PIN diode) to adjust the dual-watch balance where an attenuation of approximately 70dB (maximum) is assured. The receive level is adjustable for main band and sub band by changing the balance.

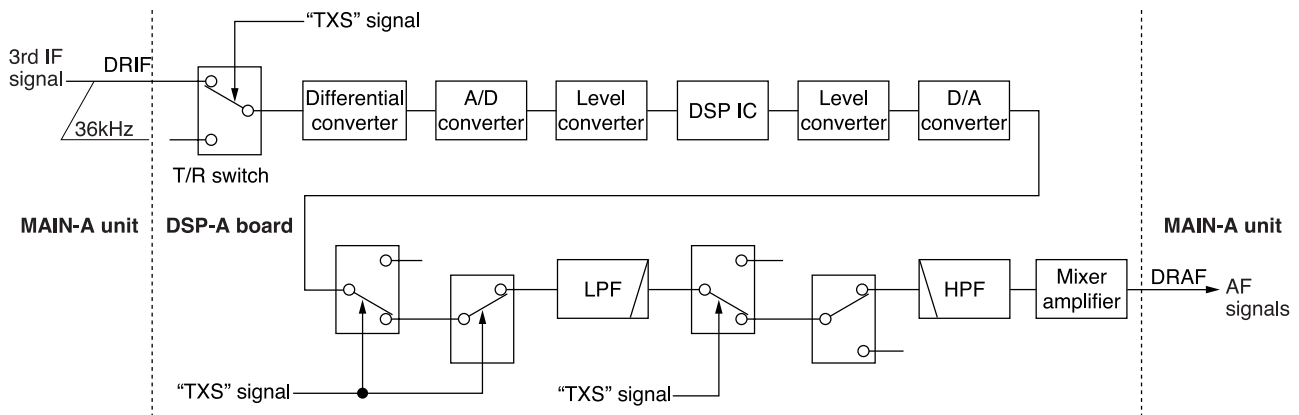
The GG amplifier (located in the succeeding stage) as well as the GG amplifier (located at the input side of the mixer) isolates the main mixer from the sub-mixer, improving the 2-signal characteristics, while maintaining the impedance (as viewed from the mixer) at a constant level. A combiner transformer determines the output for main mixer and sub mixer. The IF stage following the combiner transformer uses the circuit used for the main mixer and sub-mixer in common. The received signal passes through the 1st filter to eliminate unwanted signal components in the mixer stage. The 1st IF filter is a crystal filter selected taking 3rd order distortion into consideration. After passing through the 1st IF filter the signal is controlled by the AGC. It then enters the 2nd mixer through the 1st IF amplifier. This mixer is a diode double-balanced type with high IP which is highly effective against in-band IMD and adjacent signal interference. The element of the signal converted to 455kHz by the 2nd mixer enters the noise blanker circuit. The IF amplifier is connected to the noise blanker circuit by 4 stages in series to assure high gain. When the threshold level of the circuit used to control the noise blanker gate is varied, it is possible to change the noise blanker level in 100 stages.

The signal is further amplified by the 2nd IF amplifier and enters the 2nd IF filter. This is a ceramic filter with a high shape factor and a center frequency of 455kHz to restrict the maximum passband width of the signal passed to the DSP. The 455kHz signal is then passed to the 3rd mixer. The IC-756PROII uses a high-speed analog switch instead of the conventional mixing IC to improve the adjacent dynamic range characteristics and to restrict distortion.

An active LPF (consisting mainly of an operating amplifier) is included to collect the necessary frequency component (36kHz) from the 3rd mixer output. The capacitor of this active LPF circuit is a film type capacitor with limited distortion and superior temperature characteristics. The signal is then amplified and passed to the DSP port.

The 36kHz IF signal is differentially converted by the operating amplifier and is passed to the A/D converter. The signal is passed to the DSP IC through the level converter. The DSP IC is operated as a digital IF filter of 36kHz or as a demodulator under each mode. The demodulated signal is then passed to the D/A converter through a level converter and converted into an analog signal to pass through the low-pass filter via a differential input type active filter, buffer amplifier and analog switch to remove unwanted signals. The filtered signal passes through the analog switch to absorb the demodulation level difference between each mode with a demodulation level equalizing circuit.

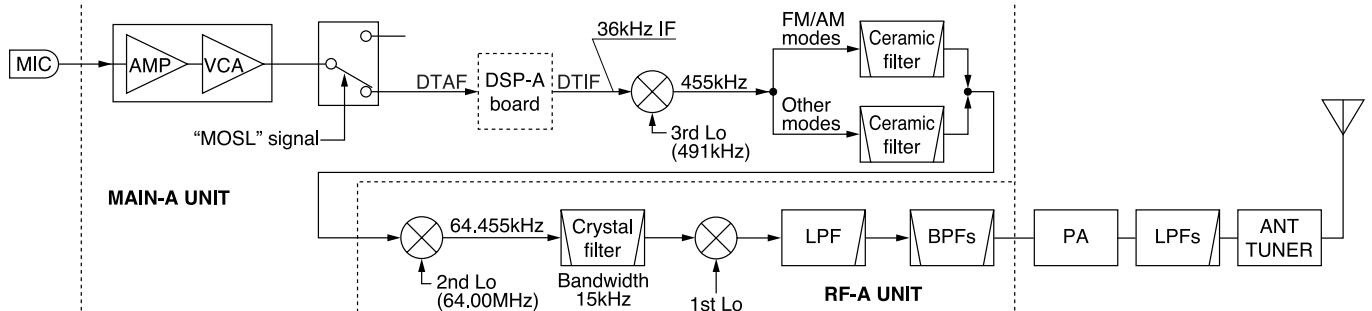
#### DSP-A board block diagram



## 5-9 Transmitter

The voice signal enters through the microphone and is amplified by the VCA (voltage control amplifier). The voice signal is controlled in gain and passed to the DSP as the DTAF signal through the analog switch. The VCA controls the gain of the microphone in accordance with a signal output from the main CPU. When SSB mode is selected, the signal enters the amplifier through the analog switch and passes through the low-pass filter entering the differential amplifier, to restrict the band of the A/D converter input signal. When FM/AM mode is selected, the signal passes through the limiter amplifier, low-pass filter and pre-emphasis circuit, and enters the differential amplifier in the same manner as SSB mode. The amplified signal enters the A/D converter and enters the DSP IC through the level converter. After the signal has been demodulated by the DSP IC, it is output as a 36kHz transmit IF signal. The demodulated signal passes through the level converter and is converted to an analog signal by the D/A converter. The analog signal passes through the differential input type active filter and enters the analog switch through the buffer amplifier.

### Transmitter block diagram



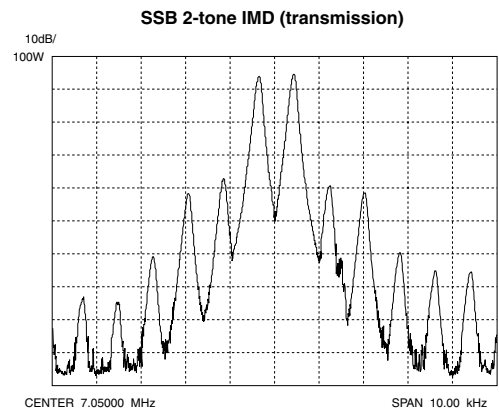
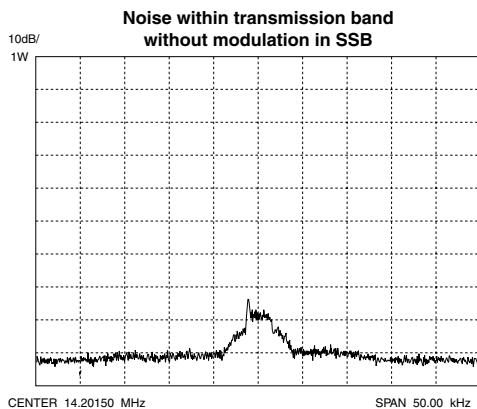
The IC-756PROII uses a well-balanced push-pull amplifier and LPF to provide an enhanced harmonic level for all bands of approximately 60dB (practical value).

The demodulation input/output to/from DSP uses the 24-bit A/D-D/A. The demodulation input/output to and from the DSP uses a 24-bit A/D-D/A converter. The use of the high-bit A/D-D/A converter significantly reduces modulation distortion due to bit error. Note that the limited number of bits causes the level deviation/bit to be increased and consequently causes the non-linear movement and demodulation distortion to be increased. The limited number of bits may also cause the maximum output level/noise output level ratio to

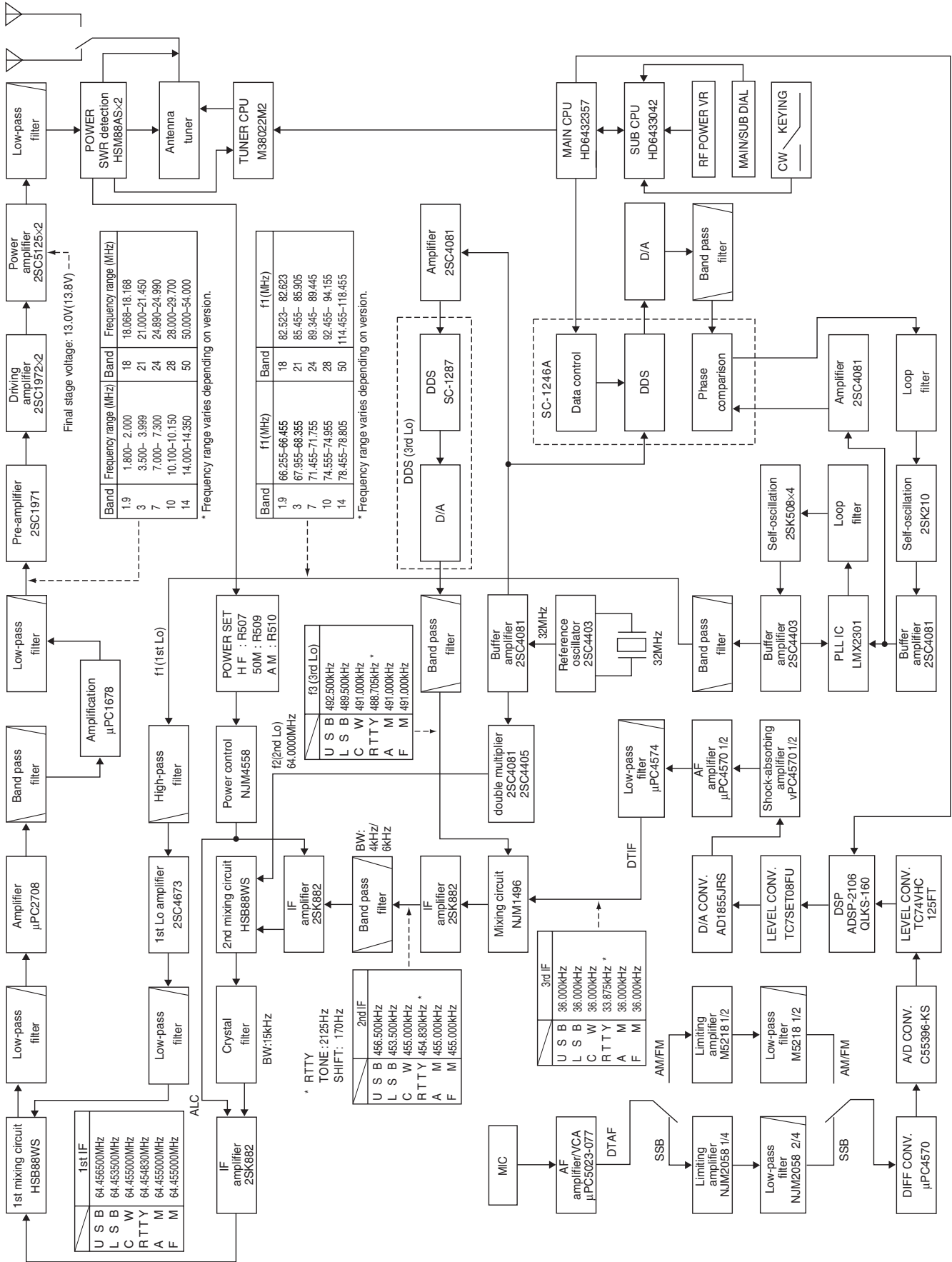
The signal then leaves the analog switch and enters the Main-A unit through the LPF as the DTIF signal to attenuate the out-of-band, spurious, or image noise. The signal is converted to the 2nd IF of 455kHz by the 3rd mixer circuit built into the Main-A unit and passes through the ceramic filter and IF filter via IF amplifier, to enter the RF-A unit. The 2nd IF signal is mixed with a 64MHz signal sent from the PLL circuit by the 2nd mixer, converted to an IF signal of 64.455MHz, stripped of unwanted components by the XTAL BPF, and enters the IF amplifier. The ALC is applied to the IF amplifier.

The IF signal is converted to the desired frequency by the HSB88WS diode mixer, stripped of unwanted frequency components by the 60MHz cut-off LPF, amplified by the RF YGR amplifier, and is then output to the PA unit. The transmit signal passes through a class A type amplifier, is amplified by the class AB push-pull amplifier, and is then amplified to 100W by the final amplifier (2SC5125 × 2). In the output of the final amplifier the higher harmonic is attenuated by the transmit PF compatible with each band.

be decreased resulting in an increased noise level when demodulation is not executed. This relationship will theoretically be "number of bits × 5dB". For a 16-bit D/A converter this is a S/N of 96dB. Comparing the rated output of 100W with the noise when demodulation is not executed the S/N will be the value obtained by subtracting the gain controlled by ALC from 96dB. When the 20dB gain control is executed at 100W, using the ALC for instance, the value of 76dB (=96 - 20) will be the ratio between noise when demodulation is not executed and level at time of 100W transmit. For 24 bits this is 124dB (=24 × 5 - 20). As a result the noise of the A/D converter is reduced to a level where it is not a problem.



# Transmitter Block Diagram



## 5-10 Dual-watch function

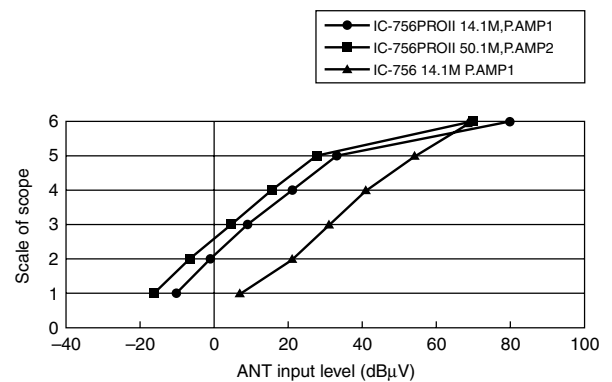
The dual-watch function allows the designated receive frequency and another frequency or the transmit/receive frequency of a DX station (used for split operation) to be watched at the same time. Fitting the transceiver with two receive circuits is one way of obtaining this dual-watch function. If a sub-receive circuit is used the performance is liable to decrease compared with the receive performance of the main receive circuit. The IC-756PROII has a dual-watch system which exceeds those in conventional equipment. Here two sets of a PLL circuit and 1st mixer are used for the dual-watch function. Both sets are used when the first intermediate frequency is attained to provide for dual watch. The ATT circuit has a PIN diode (used to adjust the balance so that the weak signal is not masked) when the

difference in the levels of the signals received is increased right after the output from the 1st mixer. The transceiver is designed to receive these two signals properly by operating the balance adjustment knob located on the front panel. The transceiver uses a gate earth buffer amplifier before and after the mixer, with the FET having significant reverse isolation, so that the oscillation signal of the 1st stations (2 sets) will not be mixed with each other. The receive characteristics of this system when the main receive circuit receives a signal is the same as when the sub-receive circuit receives a signal. This makes it possible to validate the noise reduction, noise blanker, etc. even when a signal is received by the sub-receive circuit.

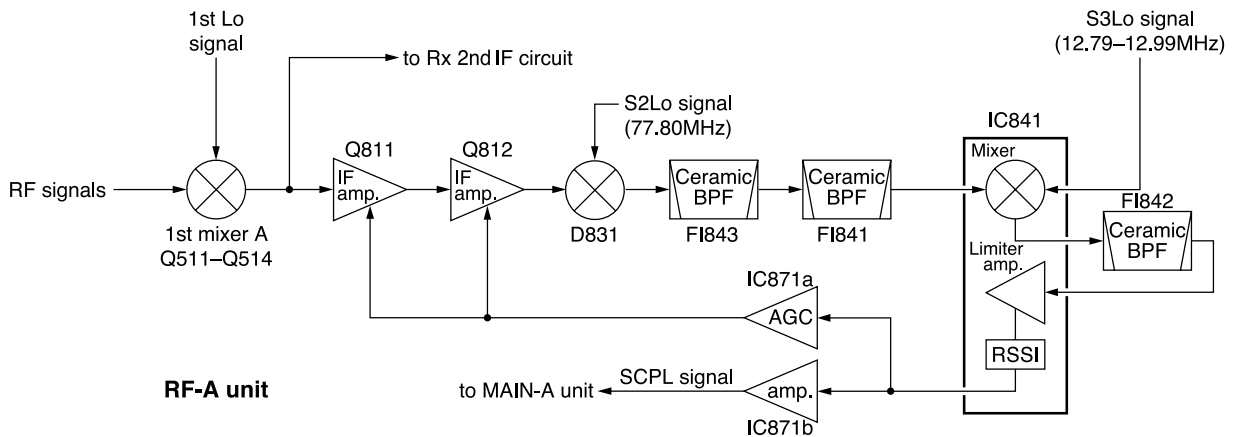
## 5-11 Real-time spectrum scope

The receive system circuit is separate from the real-time spectrum scope in the IC-756PROII. The circuit dedicated to the spectrum scope is used to enhance its accuracy. This makes it possible to use an attenuator dedicated to the spectrum scope, and to reflect the desired signal clearly without the band scope being saturated when the attenuator is turned on, even while receiving a low-level signal with a high noise floor.

Comparison of characteristics of spectrum scope



### Spectrum scope block diagram



### Description of circuit

The signal from the 1st mixer (while receiving) or the mixer in the RF-A unit (while transmitting) is passed through the PIN attenuator (D801) and amplified at Q811 and Q812, and applied to the D831 mixer. The D831 converts this signal to the 13MHz band using the 2nd mixer. The converted signal passes through the BPF, which is composed of two ceramic filters to suppress unwanted signals. The signal then enters IC841. This IC is designed for FM IF and has an

algorithmic output RSSI terminal and MIX, and is operated by sweeping the LO input to this point. The FI842 filter determines the resolution of the spectrum scope, using a ceramic filter in CW mode, to assure stable performance without need for adjustment. The RSSI voltage output from IC841 is amplified by IC871 to provide both a scope voltage and to apply the AGC to Q811 and Q812, to extend the dynamic range of the spectrum scope.

## 5-12 Voice record/playback function

This radio uses a dedicated IC (ISD4003-04) for voice recording/playback, and stores the analog signal as an analog value. Usually an analog signal is digitized temporarily and stored as a numerical value, to be converted back into an analog signal when it is reproduced. With this method it is necessary to use expensive A/D and D/A converters and storage media (RAM).

The IC-756PROII does not require these devices as it uses an IC dedicated to voice recording/playback which provides full quality audio reproduction. The previous IC-756PRO also used this method. The storage chip in the IC-756PROII was changed to one with a greater memory capacity to allow continuous recording capability.

## 5-13 PLL circuit

Since the IC-756PROII is equipped with a Dual-watch function, two sets of PLL circuits with the same configuration are included. Unlike the PLL circuits of other HF transceivers, no mixer is used. The mixer is intended to create a sum or difference for the two signals to pick out the desired frequency component. In fact the input signal as well as the sum and difference are output for mixer output. For this reason, it is necessary to arrange a filter for the mixer output. The unwanted components required by the mixer are also spurious for transmit/receive. For a system configuration in which reference oscillation is controlled by the DDS (Direct Digital Synthesizer) to output 10MHz, 10MHz is oscillated by the VCO of PLL on the basis of the 10MHz generated by this DDS. The PLL oscillating the desired frequency at 10MHz created by the VCO is controlled to oscillate the VCO. In this case, it becomes unnecessary to arrange the mixer in the transmit system.

This successful design concept, using high-speed DDS, was adopted for the first time by the IC-775DSP.

### Reference oscillator circuit

The reference oscillator circuit generates the frequency used as the reference by all oscillation circuits concerned with transmit/receive frequency. Thus the frequency accuracy depends on the accuracy of this oscillation circuit. The oscillator used for the reference oscillation circuit of the IC-756PROII provides the high accuracy of  $< -0.5\text{ppm}$

( $0.5 \times 10^{-6}$ ) when at temperatures between  $-30^{\circ}\text{C}$  to  $+60^{\circ}\text{C}$ . This oscillator is a crystal oscillator called POC, in which the oscillator is thermally balanced due to the heat generated in the resistor in the oscillator, which does not allow the temperature change to occur in the oscillator even if an external temperature change occurs. It is also possible to adjust the deviation generated due to deterioration over time.

### Configuration of LO for split operation

The 1st transmit LO for split operation is generated by changing the oscillation frequency of the PLL that generates the 1st receive LO. With this configuration no transmit or receive signals will leak at the receive frequency during split transmission.

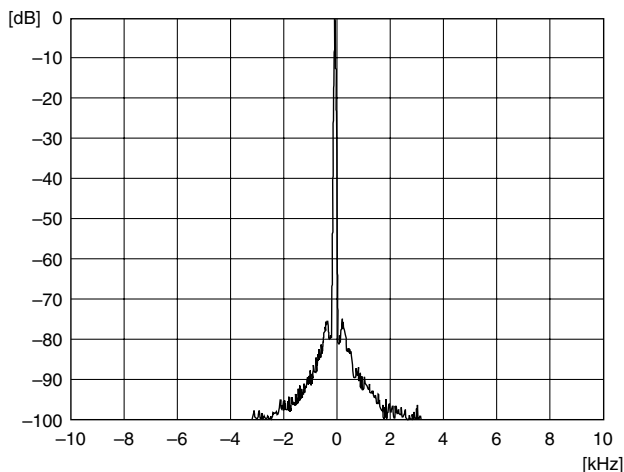
### VCO of PLL for 1st LO

Four VCOs cover the receive frequency range from 0.03MHz to 60MHz:

0.03–7.999MHz	VCO1
8–19.999MHz	VCO2
20–44.999MHz	VCO3
45–60MHz	VCO4

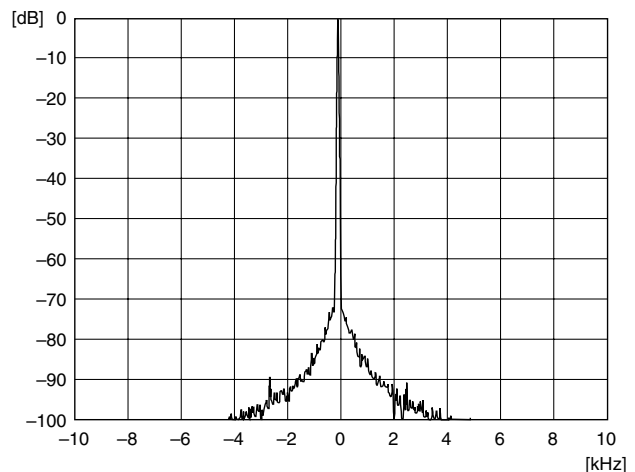
This VCO uses a HI-Q coil to minimize the noise generation, thus assuring high C/N characteristics. (C/N characteristics diagram)

1st Lo C/N characteristics of IC-756PROII



The above graphs show the 1st LO C/N characteristics for IC-756PROII and the PLL of a high grade HF unit from another manufacturer. The graph to the left indicates the LO C/N characteristic for the IC-756PROII. While the difference seems to be

1st Lo C/N characteristic of high-grade HF radio of a competitor

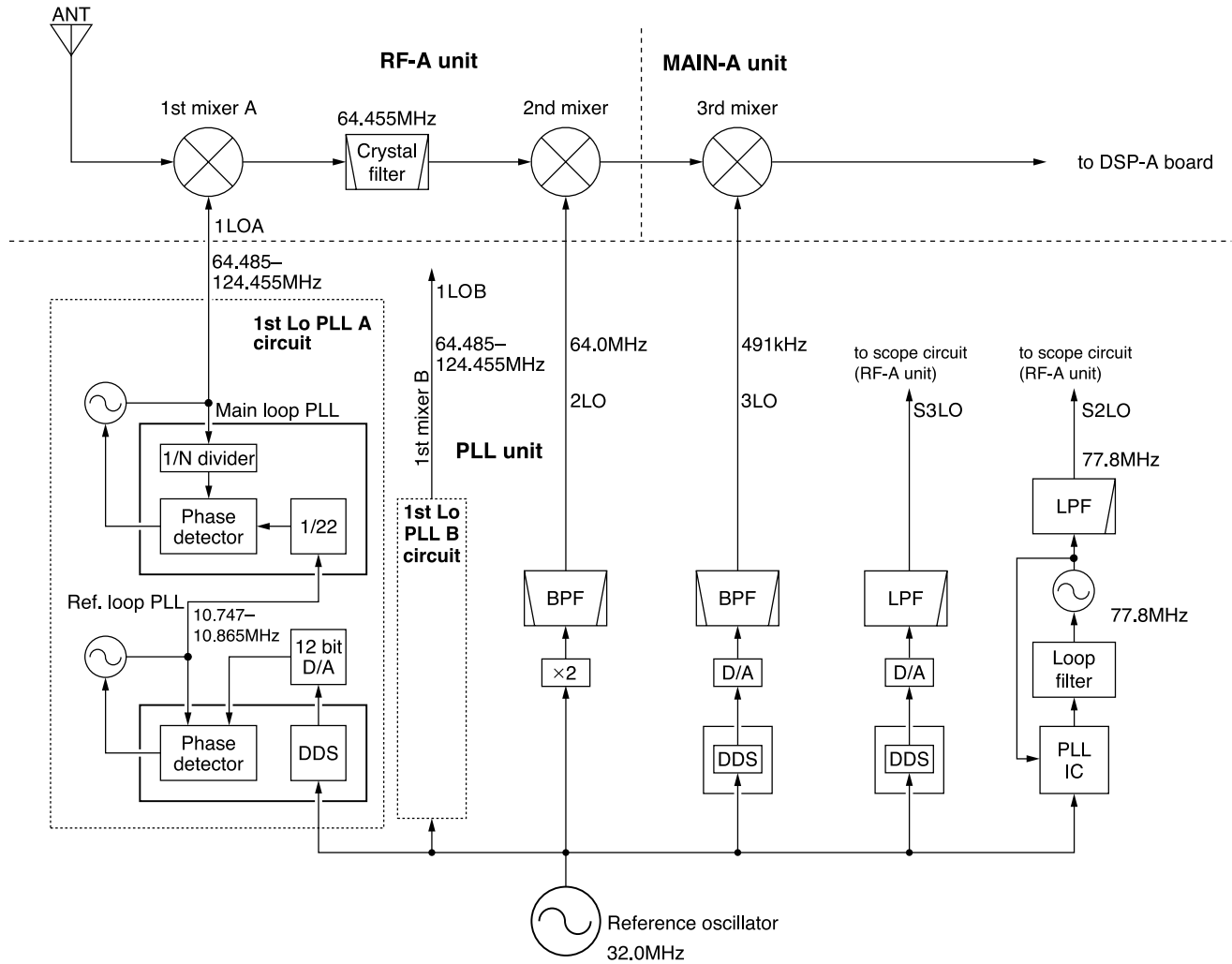


slight, the output of the 1st LO significantly affects the transmit/receive performance. When the low-band signal in the HF band is received, the difference will be apparent.

### Other LOs

The 2nd LO works to double the output of the reference oscillator circuit previously described, and the 3rd LO is obtained directly from the DDS operating in accordance with the output of the 2nd LO reference oscillator circuit. Since the PLL is not used for such frequency components, high purity and stable operation is obtained.

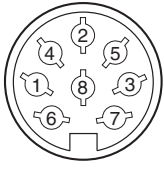
### Block diagram of PLL

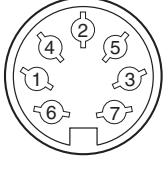




## 6. Connection to option/peripheral units

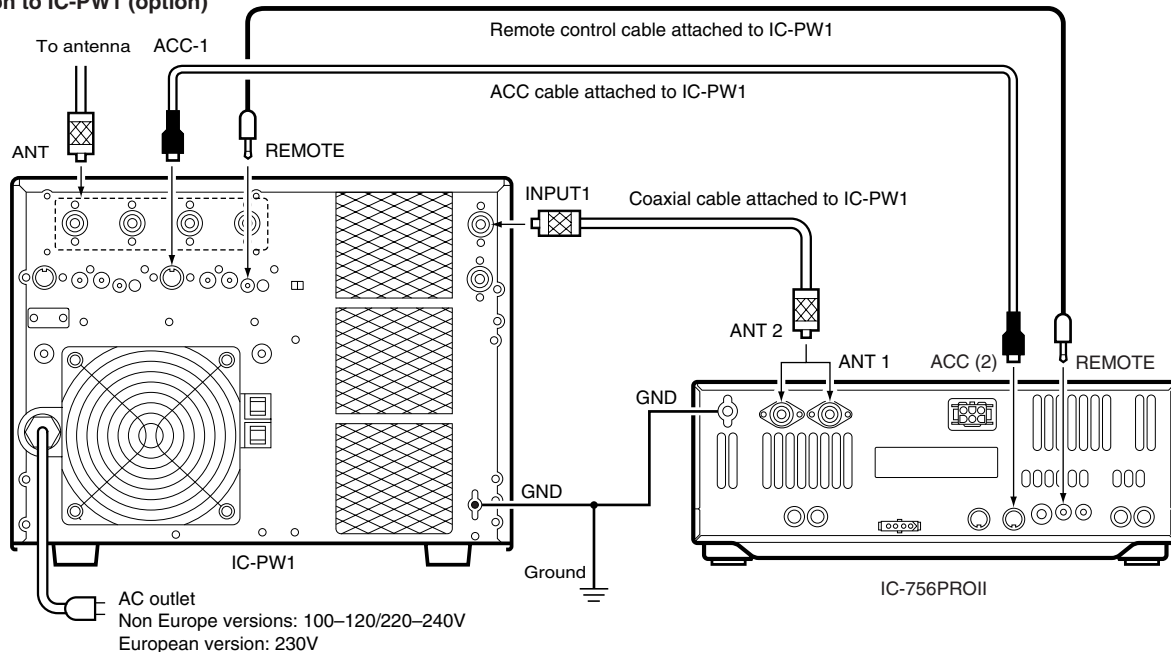
### 6-1 ACC Sockets

ACC (1)	PIN No.	NAME	DESCRIPTION	SPECIFICATIONS
 <p>Rear panel view</p>	1	RTTY	Controls RTTY keying	"High" level : More than 2.4V "Low" level : Less than 0.6V Output current : Less than 2mA
	2	GND	Connects to ground.	Connected in parallel with ACC (2) pin 2.
	3	SEND	Inout/output pin. Goes to ground when transmitting. When grounded, transmits.	Ground level : -0.5V to 0.80V Output current : Less than 200mA Input current (Tx) : Less than 200mA Connected in parallel with ACC (2) pin 3.
	4	MOD	Modulator input. Connects to a modulator.	Input impedance : 10kΩ Input level : Approx. 100mV rms
	5	AF	AF detector output. Fixed, regardless of [AF] position in default settings.	Output impedance : 4.7kΩ Output level : 100–300mV rms
	6	SQLS	Squelch output. Goes to ground when squelch opens.	SQL open : Less than 0.3V/5mA SQL closed : More than 6.0V/100μA
	7	13.8V	13.8V output when power is ON.	Output current : Max. 1 A Connected in parallel with ACC (2) pin 7.
	8	ALC	ALC voltage input.	Control voltage : -4V to 0V Input impedance : More than 10kΩ Connected in parallel with ACC (2) pin 5.

ACC (2)	PIN No.	NAME	DESCRIPTION	SPECIFICATIONS
 <p>Rear panel view</p>	1	8V	Regulated 8V output.	Output voltage : 8V ± 0.3V Output current : Less than 10mA
	2	GND	Same as ACC (1) pin 2.	
	3	SEND	Same as ACC (1) pin 3.	
	4	BAND	Band voltage output. (Varies with amateur band)	Output voltage : 0 to 8.0V
	5	ALC	Same as ACC (1) pin 8.	
	6	TRV	Activates [XVERT] input/output when "HIGH" voltage is applied.	Input impedance : More than 10kΩ Input voltage : 2 to 13.8V
	7	13.8V	Same as ACC (1) pin 7.	

### 6-2 HF/50MHz, 1kW linear amplifier

#### Connection to IC-PW1 (option)



### 6-3 Interface for digital mode

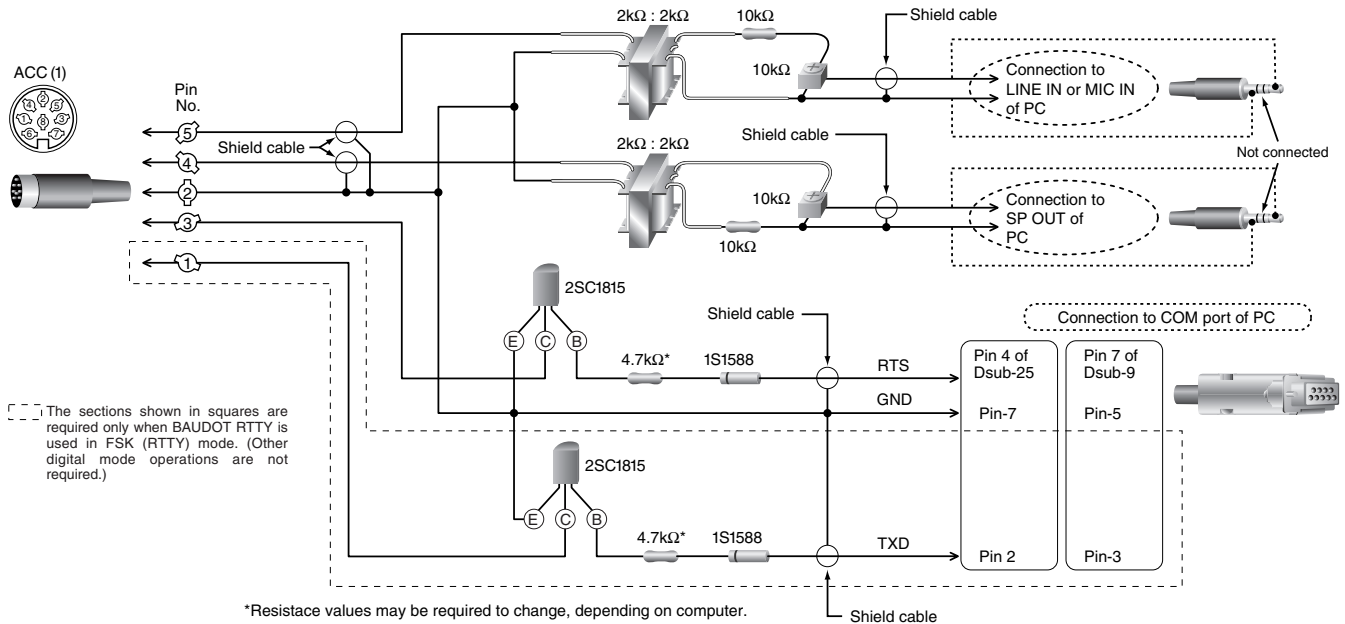
To use a personal computer to operate the digital modes (SSTV, PSK31, BAUDOT RTTY, etc.), it is necessary to install the following interface.

The IC-756PROII is equipped with a digital IF filter that may narrow the receive passband range to 50Hz making it possible to select and receive only one station, even when it is used in PSK31 mode.

If a filter width of 500Hz or less is selected the receive passband filter is activated to avoid interference while the transceiver receives SSB-D (SSB data mode).

Refer to the instruction manual or help file contained in the 3rd party software prior to use.

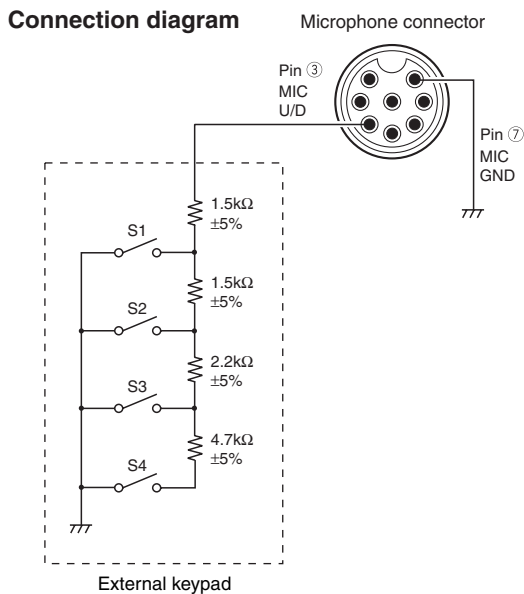
#### Example of interface for digital mode (Not provided by Icom)



This circuit connects the input and output through transformers to prevent RF feedback and to isolate the transceiver from the computer.

### 6-4 External control unit for voice memory keyer

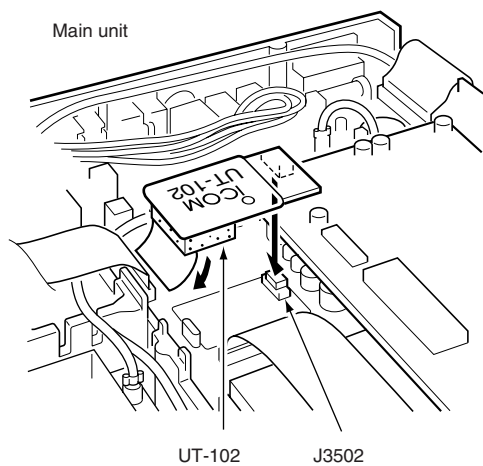
#### Example of external control circuit (Not provided by Icom.)



When a properly constructed control circuit is connected to the microphone connector, it is possible to control the transmission of the CW memory keyer (M1 to M4) and DVR (T1 to T4).

This also makes it possible to transmit the memory keyer and voice memory while displaying the scope.

## 6-5 Installation of UT-102 optional Voice Synthesizer Unit

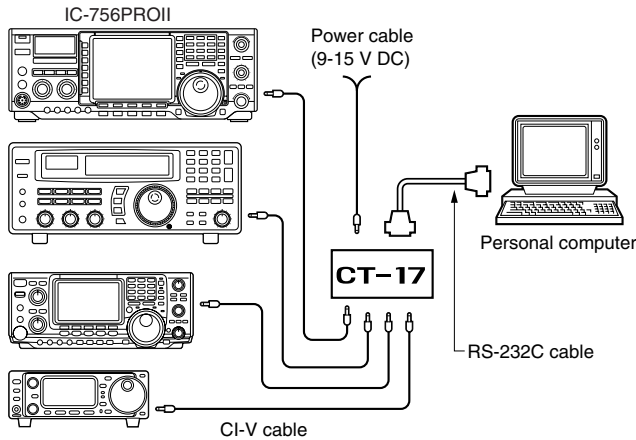


The UT-102 is capable of announcing S-meter level, frequency, and operating mode in English (on Japanese). It is possible to select voice speed (fast/slow).

## 7. CI-V control

### 7-1 Remote jack

#### Connection of computer



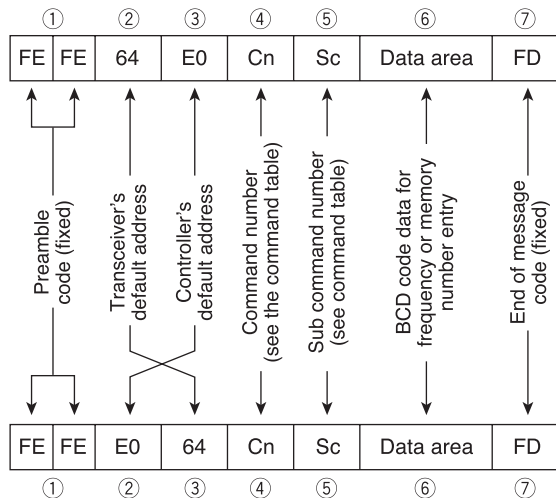
A personal computer may be used to control the frequency, operating mode, VFO/memory status, etc. via its serial port, using the Icom Communication Interface V (CI-V).

When the optional CT-17 (CI-V level converter) is connected, it is possible to control up to 4 Icom Transceivers Receivers with a personal computer. A wide variety of 3rd party software applications may be used to provide automated logging and control of your radio.

It is possible to connect up to 4 computers.

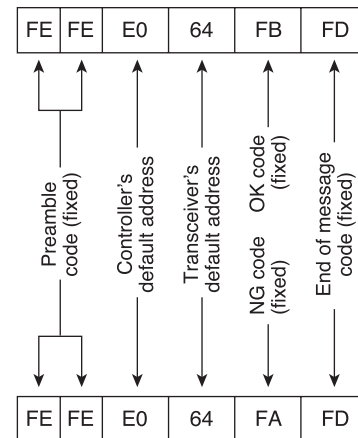
### 7-2 Data format of CI-V

#### Controller to IC-756PROII



#### IC-756PROII to controller

#### OK message to controller



#### NG message to controller

- ① Pre-amble synchronous code to insert the data at first.  
The hexadecimal "FE" is transmitted twice.
- ② Reception address
- ③ Transmission address } : The address of IC-756PROII is "64" (hexadecimal), and shows when the controller is set to "E0".
- ④ Command : The controllable function is given by a 2-digit hexadecimal command.
- ⑤ Sub-command : A 2-digit hexadecimal command is used for supplementary command instructions
- ⑥ Data area : The area is used to set the frequency data, etc., and the length is variable, depending on the data.
- ⑦ Post-amble : This is a code indicating the end of a message, and is a hexadecimal "FD".

## 7-3 List of commands

### • Command table

Command	Sub command	Description
00	–	Send frequency data
01	Same as command 06	Send mode data
02	–	Read band edge frequencies
03	–	Read operating frequency
04	–	Read operating mode
05	–	Set frequency data
06	00	Select LSB
	01	Select USB
	02	Select AM
	03	Select CW
	04	Select RTTY
	05	Select FM
	07	Select CW-R
	08	Select RTTY-R
07	–	Select VFO mode
	B0	Exchange main and sub readouts
	B1	Equalize main and sub readouts
	C0	Turn the dualwatch OFF
	C1	Turn the dualwatch ON
	D0	Select main readout
08	D1	Select sub readout
	–	Select memory mode
	0001–0101*1	Select memory channel *1P1=0100, P2=0101
09	–	Memory write
0A	–	Memory to VFO
0B	–	Memory clear
0E	00	Scan stop
	01	Programmed/memory scan start
	02	Programmed scan start
	03	$\Delta$ F scan start
	12	Fine programmed scan start
	13	Fine $\Delta$ F scan start
	22	Memory scan start
	23	Select memory scan start
	A1–A7	Set $\Delta$ F scan span (A1= $\pm$ 5kHz, A2= $\pm$ 10kHz, A3= $\pm$ 20kHz, A4= $\pm$ 50kHz, A5= $\pm$ 100kHz, A6= $\pm$ 500kHz, A7= $\pm$ 1MHz)
	B0	Set as non-select channel
	B1	Set as select channel
	D0	Set scan resume OFF
	D3	Set scan resume ON
0F	00	Turn the split function OFF
	01	Turn the split function ON
10	00	Select 10Hz (1Hz) tuning step
	01	Select 100Hz tuning step
	02	Select 1kHz tuning step
	03	Select 5kHz tuning step
	04	Select 9kHz tuning step
	05	Select 10kHz tuning step
	06	Select 12.5kHz tuning step
	07	Select 20kHz tuning step
08	Select 25kHz tuning step	
11	00	Attenuator OFF
	06	Attenuator ON (6dB)
	12	Attenuator ON (12dB)
	18	Attenuator ON (18dB)

Command	Sub command	Description
12	00	Select/read antenna selection (00=ANT1, 01=ANT2 : Add 0 or 1 to turn [RX ANT ] OFF or ON, respectively.)
	01	
13	00	Announce with voice synthesizer (00=all data; 01=frequency and S-meter level; 02=receive mode )
	01	
	02	
14	01 + Level data	[AF] level setting (0=max. CCW to 255=max. CW)
	02 + Level data	[RF] level setting (0=max. CCW to 255=11 o'clock)
	03 + Level data	[SQL] level setting (0=11 o'clock to 255=max. CW)
	06 + Level data	[NR] level setting (0=min. to 255=max.)
	07 + Level data	Inside [TWIN PBT] setting or IF shift setting (0=max. CCW, 128=center, 255=max. CW)
	08 + Level data	Outside [TWIN PBT] setting (0=max. CCW, 128=center, 255=max.CW)
	09 + Level data	[CW PITCH] setting (0=low pitch to 255=high pitch)
	0A + Level data	[RF POWER] setting (0=mini. to 255=max.)
	0B + Level data	[MIC GAIN] setting (0=mini. to 255=max.)
	0C + Level data	[KEY SPEED] setting (0=slow to 255=fast)
	0D + Level data	[NOTCH] setting (0=low freq. to 255=high freq.)
	0E + Level data	[COMP] setting (0=mini. to 255=max.)
	0F + Level data	[BK-IN DELAY] setting (0=short delay to 255=long delay)
10 + Level data	[BAL] level setting (0=max. CCW, 128=center, 255=max. CW)	
15	01	Read squelch condition
	02	Read S-meter level
16	02	Preamp (0=OFF; 1=preamp 1; 2=preamp 2)
	12	AGC selection (1=Fast; 2=Mid; 3=Slow)
	22	Noise blanker (0=OFF; 1=ON)
	40	Noise reduction (0=OFF; 1=ON)
	41	Auto notch (0=OFF; 1=ON)
	42	Repeater tone (0=OFF; 1=ON)
	43	Tone squelch (0=OFF; 1=ON)
	44	Speech compressor (0=OFF; 1=ON)
	45	Monitor(0=OFF; 1=ON)
	46	VOX function (0=OFF; 1=ON)
47	Break-in (0=OFF; 1=semi break-in; 2=full break-in)	
48	Manual notch (0=OFF; 1=ON)	
49	RTTY filter (0=OFF; 1=ON)	
19	00	Read the transceiver ID
1A	00	Send/read memory contents (see p. 31 for details)
	01	Send/read band stacking register contents (see p. 31 for details)
	02	Send/read memory keyer contents (see p. 31 for details)
	03	Send/read the selected filter width (0=50Hz to 40/31=3600/2700Hz)

• Command table (continued)

Command	Sub command	Description
1A	04	Send/read the selected AGC time constant (0=OFF, 1=0.1/0.3 sec. to 13=6.0/8.0 sec.)
	0501	Send/read SSB TX Tone (Bass) level (0=min. to 10=max.)
	0502	Send/read SSB TX Tone (Treble) level (0=min. to 10=max.)
	0503	Send/read MONITOR gain (0=min. to 255=max.)
	0504	Send/read CW side tone gain (0=min. to 255=max.)
	0505	Send/read CW side tone gain limit (0=OFF, 1=ON)
	0506	Send/read beep gain (0=min. to 255=max.)
	0507	Send/read beep gain limit (0=OFF, 1=ON)
	0508	Send/read LCD contrast (0=0% to 255=100%)
	0509	Send/read LCD Backlight (0=0% to 255=100%)
	0510	Send/read LCD horizontal position (0=1 to 7=8)
	0511	Send/read switch backlight (0=1 to 7=8)
	0512	Send/read display type (0=A, 1=B, 2=C, 3=D, 4=E, 5=F, 6=G, 7=H)
	0513	Send/read display font (0=Basic1, 1=Basic2, 2=Pop, 3=7seg, 4=Italic1, 5=Italic2, 6=Classic)
	0514	Send/read memory name (0=OFF, 1=ON)
	0515	Send/read my call setting (10-character: see p. 31)
	0516	Send/read current time (0000 to 2359)
	0517	Send/read power-ON timer set (0000 to 2359)
	0518	Send/read power-OFF period (5=5 min. to 120=120 min. in 5 min. step)
	0519	Send/read calibration marker (0=OFF, 1=ON)
	0520	Send/read confirmation beep (0=OFF, 1=ON)
	0521	Send/read band edge beep (0=OFF, 1=ON)
	0522	Send/read RF/SQL control set (0=Auto, 1=SQL, 2=RF+SQL)
	0523	Send/read quick dualwatch set (0=OFF, 1=ON)
	0524	Send/read quick split set (0=OFF, 1=ON)
	0525	Send/read FM split offset (HF) -4,000 to + 4,000MHz (see p. 31 for details)
	0526	Send/read FM split offset (50MHz) -4,000 to + 4,000MHz (see p. 31 for details)
	0527	Send/read split lock set (0=OFF, 1=ON)
	0528	Send/read tuner auto start set (0=OFF, 1=ON)
	0529	Send/read PTT tune set (0=OFF, 1=ON)
0530	Send/read antenna selection (0=OFF, 1=Manual, 2=Auto)	

Command	Sub command	Description
1A	0531	Send/read RTTY mark frequency (0=1275Hz, 1=1615Hz, 2=2125Hz)
	0532	Send/read RTTY shift width (0=170Hz, 1=200Hz, 2=425Hz)
	0533	Send/read RTTY keying polarity (0=Normal, 1=Reverse)
	0534	Send/read RTTY decode USOS (0=OFF, 1=ON)
	0535	Send/read RTTY decode new line code (0=CR, LF, CR+LF, 1=CR+LF)
	0536	Send/read speech language (0=English, 1=Japanese)
	0537	Send/read speech speed (0=slow, 1=fast)
	0538	Send/read S-level speech (0=OFF, 1=ON)
	0539	Send/read memo pad numbers (0=5 ch, 1=10 ch)
	0540	Send/read main dial auto TS (0=OFF, 1=Low, 2=High)
	0541	Send/read mic. up/down speed (0=Low, 1=High)
	0542	Send/read CI-V transceive set (0=OFF, 1=ON)
	0543	Send/read CI-V 731 mode set (0=OFF, 1=ON)
	0544	Send/read TX spectrum scope set (0=OFF, 1=ON)
	0545	Send/read spectrum scope max. hold set (0=OFF, 1=ON)
	0546	Send/read voice auto monitor set (0=OFF, 1=ON)
	0547	Send/read cut number style (0=Normal, 1=190→ANO, 2=190→ANT, 3=90→NO, 4=90→NT)
	0548	Send/read count up trigger channel (1=M1, 2=M2, 3=M3, 4=M4)
	0549	Send/read present number (1-9999)
	0550	Send/read CW keyer repeat time (1=1 sec. to 60=60 sec.)
	0551	Send/read CW keyer dot/dash ratio (28=1:1.2:8 to 45=1:1:4.5)
	0552	Send/read rise time (0=2 msec., 1=4 msec., 2=6 msec., 3=8 msec.)
	0553	Send/read paddle polarity (0=Normal, 1=Reverse)
	0554	Send/read keyer type (0=Straight, 1=Bug-key, 2=ELEC-Key)
	0555	Send/read mic. up/down keyer set (0=OFF, 1=ON)
	0556	Send/read scan speed (0=low, 1=high)
	0557	Send/read scan resume (0=OFF, 1=ON)
	0558	Send/read VOX gain (0=0% to 255=100%)
	0559	Send/read anti VOX gain (0=0% to 255=100%)
	0560	Send/read VOX delay (0=0.0 sec. to 20=2.0 sec.)
0561	Send/read RTTY filter bandwidth (0=250Hz, 1=300Hz, 2=350Hz, 3=500Hz, 4=1kHz)	

• **Command table (continued)**

Command	Sub command	Description
1A	0562	Send/read twin peak filter (0=OFF, 1=ON)
	0563	Send/read timer functions (0=OFF, 1=ON)
	0564	Send/read DSP filter type (0=SSB: sharp; CW: sharp, 1=SSB: sharp; CW: soft, 2=SSB: soft CW: sharp, 3=SSB: soft CW: soft)
	0565	Send/read quick RIT/ $\Delta$ TX clear function (0=OFF, 1=ON)
	0566	Send/read SSB/CW synchronous tuning function (0=OFF, 1=ON)
	0567	Send/read CW normal side set (0=LSB, 1=USB)
	0568	Send/read external keypad type (0=OFF, 1=Keyer send, 2=Voice play (Tx), 3=Auto)
	0569	Send/read NB level (0=0% to 255=100%)
	06	Send/read DATA mode (0=OFF, 1=ON)
	07	Send/read SSB transmit bandwidth (0=Wide, 1=Middle, 2=Narrow)
1B	00	Set repeater tone frequency
	01	Set tone squelch tone frequency
1C	00	Set the transceiver to receive or transmit condition (0=Rx; 1=Tx)

• **To send/read memory contents**

When sending or reading memory contents, additional code as follows must be added to appoint the memory channel.

➔ Additional code: 0000–0101 (0100=P1, 0101=P2)

• **Band stacking register**

To send or read desired band stacking register's contents, combined code of the frequency band and register codes as follows are used.

For example, when sending/reading the oldest contents in the 21 MHz band, the code "0703" is used.

• **Frequency band code**

Code	Frequency band	Frequency range (unit: MHz)
01	1.8	1.800000– 1.999999
02	3.5	3.400000– 4.099999
03	7	6.900000– 7.499999
04	10	9.900000–10.499999
05	14	13.900000–14.499999
06	18	17.900000–18.499999
07	21	20.900000–21.499999
08	24	24.400000–25.099999
09	28	28.000000–29.999999
10	50	50.000000–54.000000
11	GENE	Other than above

• **Register code**

Code	Registered number
01	1 (latest)
02	2
03	3 (oldest)

• **Channel code for memory keyer**

To send or read the desired memory keyer contents, the channel and character codes as follows are used.

• **Channel code**

Code	Channel number
01	M1
02	M2
03	M3
04	M4

• **Character's code**

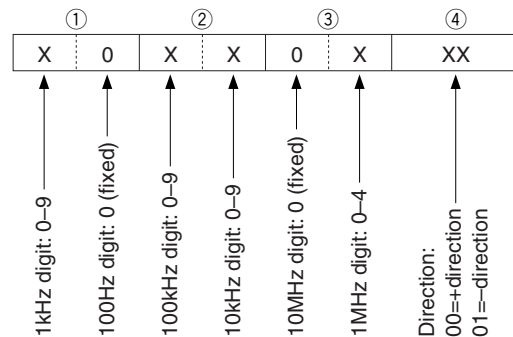
Character	ASCII code	Description
0–9	30–39	Numerals
A–Z	41–5A	Alphabetical characters
a–z	61–7A	Alphabetical characters
space	20	Word space
/	2F	Symbol
?	3F	Symbol
,	2C	Symbol
.	2E	Symbol
^	5E	e.g., to send $\overline{BT}$ , enter ^4254
*	2A	Inserts contact number (can be used for 1 channel only)

• **Character's code for my call**

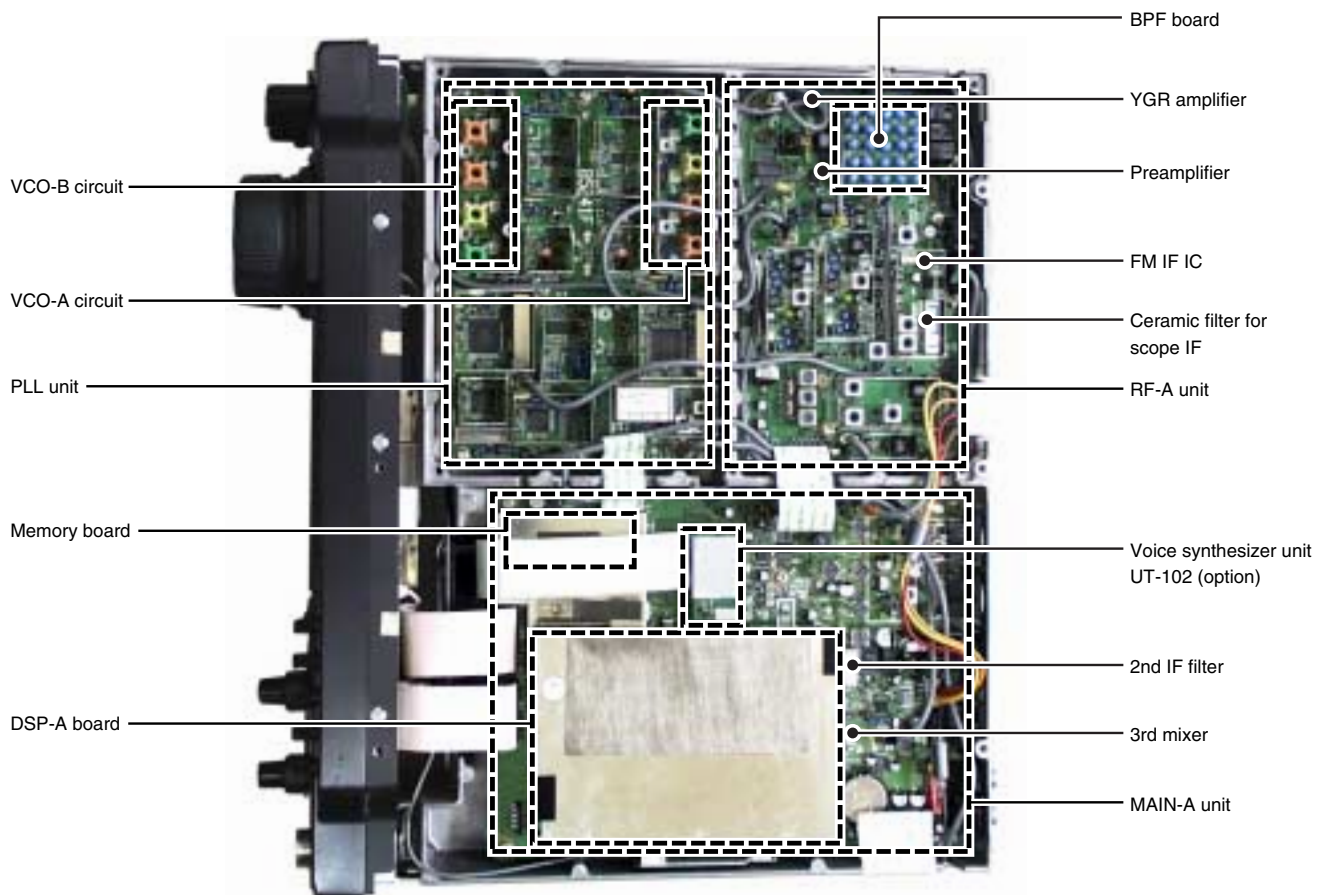
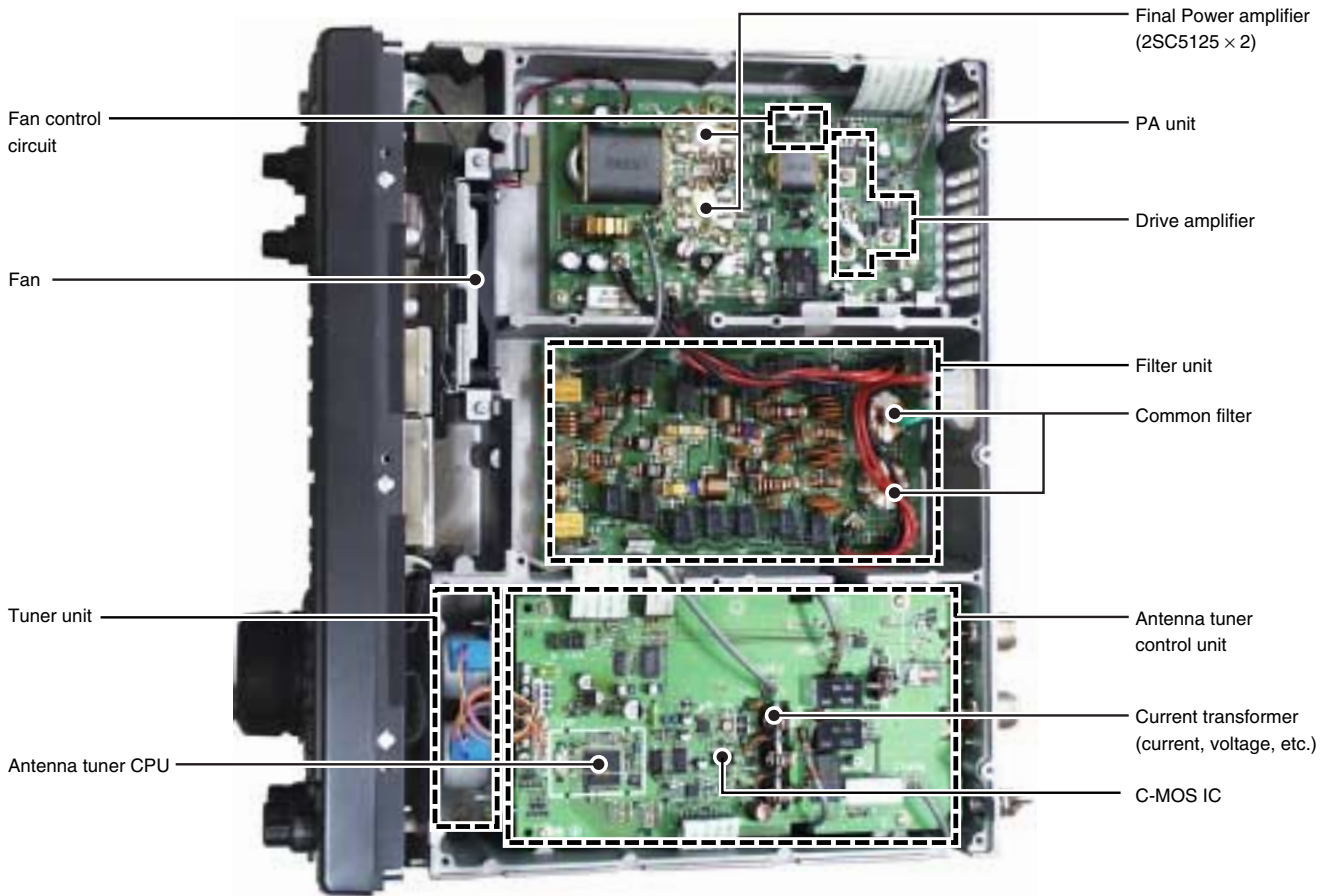
Character	ASCII code	Description
0–9	30–39	Numerals
A–Z	41–5A	Alphabetical characters
a–z	61–7A	Alphabetical characters
space	20	Word space
–	2D	Symbol
.	2E	Symbol
/	2F	Symbol

• **FM split frequency (HF/50MHz) setting**

The following data sequence is used when sending/reading the FM split frequency setting.



## 8. Inside Views





## 9. Options

### IC-PW1 HF/50MHz ALL BAND 1kW LINEAR AMPLIFIER



Full-duty 1kW linear amplifier including an automatic antenna tuner. Has automatic tuning and band selection capability. Full break-in (QSK) operation is possible. The amplifier/power supply unit and the remote control unit are separated.

### PS-125 DC POWER SUPPLY



Light weight power supply.  
• Output voltage: 13.8V DC  
• Max. current drain: 25A

### AH-4 HF/50 MHz AUTOMATIC ANTENNA TUNER



Specially designed to tune a long wire antenna for portable or mobile 3.5–54MHz operation.  
• Input power rating: 120W

### AH-2b ANTENNA ELEMENT



A 2.5 m long antenna element for mobile operation with the AH-4.  
• Frequency coverage: 7–54MHz bands with the AH-4

### HM-36 HAND MICROPHONE



Hand microphone equipped with [UP]/[DOWN] switches. Same as supplied.

### SM-20 DESKTOP MICROPHONE



Unidirectional, electret microphone for base station operation. Includes [UP]/[DOWN] switches and a low cut function.

### SP-20 EXTERNAL SPEAKER



4 audio filters; headphone jack; can connect to 2 transceivers.  
• Input impedance: 8Ω  
• Max. input power: 5W

### SP-21 EXTERNAL SPEAKER



Designed for base station operation.  
• Input impedance: 8Ω  
• Max. input power: 5W

### CT-17 CI-V LEVEL CONVERTER



For remote transceiver control using a PC. You can change frequencies, operating mode, memory channels, etc.

### UT-102 VOICE SYNTHESIZER UNIT



Announces the receive frequency, mode and S-meter level in a clear, electronically-generated voice, in English (or Japanese).

## 10. Specifications

### GENERAL

- Frequency coverage :
 

U.S.A.	Rx	0.030–60.000* <sup>1</sup>	
	Tx	1.800– 2.000* <sup>1</sup>	3.500– 3.999
		7.000– 7.300	10.100–10.150
		14.000–14.350	18.068–18.168
		21.000–21.450	24.890–24.990
		28.000–29.700	50.000–54.000
Europe	Rx	0.030–60.000* <sup>1</sup>	
	Tx	1.800– 1.999	3.400– 4.099* <sup>1</sup>
		6.900– 7.499* <sup>1</sup>	9.900–10.499* <sup>1</sup>
		13.900–14.499* <sup>1</sup>	17.900–18.499* <sup>1</sup>
		20.900–21.499* <sup>1</sup>	24.400–25.099* <sup>1</sup>
		28.000–29.999* <sup>1</sup>	50.000–52.000
France	Tx/Rx	1.810– 1.850 (France)	
Italy		1.830– 1.850 (Italy, Spain)	
Spain		3.500– 3.800	7.000– 7.100
		10.100–10.150	14.000–14.350
		18.068–18.168	21.000–21.450
		24.890–24.990	28.000–29.700
		50.200–51.200 (France)	
		50.000–51.000 (Italy)	
		50.000–50.200 (Spain)	

\*<sup>1</sup> Some freq. bands are not guaranteed.
- Mode : USB, LSB, CW, RTTY, AM, FM
- Number of memory Ch. : 101 (99 regular, 2 scan edges)
- Antenna connector : SO-239×2 and phono [RCA; (50Ω)]
- Temperature range : –10°C to +50°C; +14°F to +122°F
- Frequency stability : Less than ±0.5ppm (From 1 minute after power ON at 0°C to 50°C; +32°F to +122°F)
- Frequency resolution : 1Hz
- Power supply requirement : 13.8V DC ±15% (negative ground)
- Power consumption : Tx Max. power 23A
 

Rx	Standby	3.0A (typ.)
	Max. audio	3.3A (typ.)
- Dimensions : 340(W)×111(H)×285(D) mm;  
(projections not included) 13<sup>3</sup>/<sub>8</sub>(W)×4<sup>3</sup>/<sub>8</sub>(H)×11<sup>7</sup>/<sub>32</sub>(D) in
- Weight (approx.) : 9.6kg; 21.2lb
- ACC 1 connector : 8-pin DIN connector
- ACC 2 connector : 7-pin DIN connector
- CI-V connector : 2-conductor 3.5 (d) mm (1/8")
- Display : 5-inch (diagonal) TFT color LCD

### TRANSCEIVER

- Output power : SSB, CW, RTTY, FM 5–100W  
(continuously adjustable) AM 5–40W
- Modulation system : SSB DPSN modulation  
AM Digital low power modulation  
FM Digital phase modulation
- Spurious emission : 50dB (HF bands)  
60dB (50MHz band)
- Carrier suppression : More than 40dB
- Unwanted sideband suppression: More than 55dB
- ΔTX variable range : ±9.999kHz
- Microphone connector : 8-pin connector (600Ω)
- ELE-KEY connector : 3-conductor 6.35 (d) mm (1/4")
- KEY connector : 3-conductor 6.35 (d) mm (1/4")
- SEND connector : Phono (RCA)
- ALC connector : Phono (RCA)

### RECEIVER

- Receive system : Triple conversion superheterodyne system
- Intermediate frequencies : 1st 64.455MHz (for all modes)  
2nd 455kHz (for all modes)  
3rd 36kHz (for all modes)
- Sensitivity (typical) :
 

SSB, CW, RTTY (10dB S/N)	0.16μV* <sup>1</sup> (1.80–29.99MHz)
AM (10dB S/N)	0.13μV* <sup>2</sup> (50.0–54.0MHz)
	13μV (0.5–1.799MHz)
	2μV* <sup>1</sup> (1.80–29.99MHz)
	1μV (50.0–54.0MHz)
FM (12dB SINAD)	0.5μV* <sup>1</sup> (28.0–29.9MHz)
	0.32μV* <sup>2</sup> (50.0MHz–54.0MHz)

\*<sup>1</sup>Pre-amp 1 is ON, \*<sup>2</sup>Pre-amp 2 is ON
- Squelch sensitivity (Pre-amp: OFF):
 

SSB, CW, RTTY	Less than 5.6μV
FM	Less than 1μV
- Selectivity (representative value):
 

SSB, RTTY (BW: 2.4kHz)	More than 2.4kHz/–6dB Less than 3.2kHz/–40dB Less than 3.6kHz/–60dB Less than 4.3kHz/–80dB
CW (BW: 500Hz)	More than 500Hz/–6dB Less than 700Hz/–60dB
AM (BW: 6kHz)	More than 6.0kHz/–6dB Less than 15.0kHz/–60dB
FM (BW: 15kHz)	More than 12.0kHz/–6dB Less than 20.0kHz/–60dB
- Spurious and image rejection ratio : More than 70dB  
(except IF through on 50MHz band)
- AF output power : More than 2.0W at 10% distortion  
(at 13.8V DC)
- RIT variable range : ±9.999kHz
- PHONES connector : 2-pin connector 6.35 (d) mm (1/4")
- EXT SP connector : 2-pin connector 3.5 (d) mm (1/4")/8Ω

### ANTENNA TUNER

- Matching impedance range:
 

16.7–150Ω unbalanced* <sup>1</sup> (HF bands)
20–125Ω unbalanced* <sup>2</sup> (50MHz band)

\*<sup>1</sup>Less than VSWR 3:1; \*<sup>2</sup>Less than VSWR 2.5:1
- Min. operating input power: 8 W
- Tuning accuracy : VSWR 1.5:1 or less
- Insertion loss : Less than 1.0dB (after tuning)

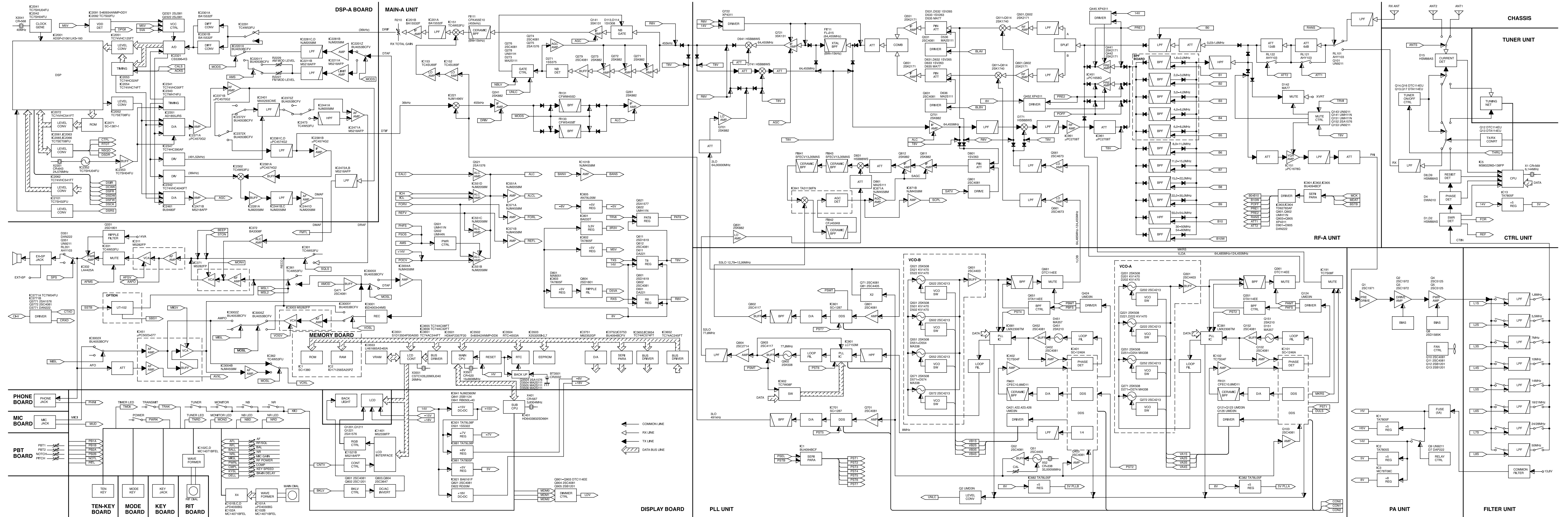
#### Supplied accessories:

- |                          |                  |
|--------------------------|------------------|
| • Hand microphone, HM-36 | • DC power cable |
| • Spare fuses            | • CW key plug    |

The LCD display may have cosmetic imperfections that appear as small or dark spots. This is not a malfunction or defect, but a normal characteristic of LCD displays. All trademarks are the properties of their respective holders.

**All stated specifications are subject to change without notice or obligation.**

# 11. Block diagram





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Your local distributor/dealer:



Certificate Number Q14190

Icom Inc. (Japan), is an ISO9001 certification acquired company.