

Type	num	Settable?	Encoding	Units	Global	Valid range/values	Meaning & Use
AD_BITS_PER_SAMPLE	82		uint	bits	Y		Width of A/D input word
AD_OVER	104		uint		Y		Count of A/D overrange events
AGC_ENABLE	62	Y	boolean				Automatic gain control (Linear demod only)
AGC_HANGTIME	64	Y	float32	sec		>= 0	Time delay before automatic gain increase on lowered signal (linear demod only)
AGC_RECOVERY_RATE	65	Y	float32	dB/sec		>= 0	Gain increase rate on lowered signal (linear demod only)
AGC_THRESHOLD	67	Y	float32	dBFS		<= 0	Target demodulator output level on noise only
BASEBAND_POWER	46		float32	dB			Signal power at channel downconverter filter output, relative to unity
BIN_BYTE_DATA	9		bytes			Limited by IP packet size	Vector representing spectrum data, 1 bin/byte, unsigned, with meaning SPECTRUM_BASE + SPECTRUM_STEP * x dB
BIN_COUNT	94	Y	uint	bins		> 0	Number of FFT bins wanted in spectrum data
BIN_DATA	96		float32 vector				Vector of 4-byte float32s with linear spectrum data, spectrum mode only. Order: DC...max positive freq, max neg freq...-1
BLOCKS_SINCE_POLL	103		uint				Count of status frames sent since last command received. Deprecated
CALIBRATE	24	Y	double64		Y		Frequency calibration factor for tuner reference and A/D sample clock. Actual freq = nominal * (1 + CALIBRATE), ie, 0 means "on frequency"
CLEAROPTS	7	Y	uint			32 bits	1-bits turn off specified option (32 max)
CMD_CNT	2		uint		Y		Server count of received commands
COHERENT_BIN_SPACING	92		float32		y		deprecated; can be calculated from INPUT_SAMPRATE, FILTER_BLOCKSIZE, FILTER_FIR_LENGTH
COMMAND_TAG	1	Y	uint		Y		generated by controller, echoed by server to confirm command
CONVERTER_OFFSET	88		float32	Hz	Y		Frequency offset of external frequency converter. Not yet implemented
CROSSOVER	95	Y	float32			positive	Value of RESOLUTION_BW above which the wideband spectrum analyzer is used.
DC_I_OFFSET	28		float32		Y		DC offset of I-channel A/D converter (only direct conversion front ends)
DC_Q_OFFSET	29		float32		Y		DC offset of Q-channel A/D converter (only direct conversion front ends)
DEEMPH_GAIN	87		float32	dB			Static gain correction when de-emphasis used to maintain subjectively equal loudness
DEEMPH_TC	86		float32	s		non-negative	Deemphasis time constant (0 = off), FM only
DEMOD_TYPE	48	Y	uint	enum		0 - 4	Demodulator type, enum: 0 = linear; 1 = FM/PM; 2 = Wideband FM with stereo demodulator; 3 = spectrum; 4 = spectrum v2
DESCRIPTION	4		string		Y	free-form UTF-8	description of front end (antenna, etc). Generated by front end, passed through 'radio'. Need not be null terminated.
DIRECT_CONVERSION	32		boolean		Y		Front end uses direct conversion with DC spike and 1/f noise that should be avoided
DOPPLER_FREQUENCY	37	Y	double64	Hz			Doppler tuning offset (untested)
DOPPLER_FREQUENCY_RATE	38	Y	double64	Hz/sec			Rate of change of Doppler tuning effort (untested). Limited to 1 bin/frame time
ENVELOPE	56	Y	boolean				Use envelope detector in linear demodulator
EOL	0						End of option list. No length or value field
FE_HIGH_EDGE	101		float32	Hz	Y	> FE_LOW_EDGE	Upper edge of A/D converter input frequency band (negative for tuners with high side injection)
FE_ISREAL	102		boolean		Y		Y: front end uses a single A/D converter; N: front end uses dual (I/Q) A/D converters
FE_LOW_EDGE	100		float32	Hz	Y	<FE_HIGH_EDGE	Low edge of A/D converter input frequency band (negative for tuners with high side injection)
FILTER_BLOCKSIZE	42		uint	samples	Y	>0	Input samples per downconverter FFT processing block
FILTER_DROPS	77		uint	blocks			Number of frame drops by digital downconverter
FILTER_FIR_LENGTH	43		uint	samples	Y	>0	Overlap samples per downconverter FFT processing block. Sets maximum impulse duration of downconverter channel filter
FILTER2	44	Y	uint	frames		0-10	Size of secondary filter input in units of downconverter frames (eg. 20 ms). 0=off
FILTER2_BLOCKSIZE	73		uint	samples		>0	Input samples per Filter2 block
FILTER2_FIR_LENGTH	74		uint	samples		>0	Impulse response length in secondary filter (filter2)
FILTER2_KAISER_BETA	75	Y	float32			>=0	Kaiser β factor for secondary filter (filter2) window design
FIRST_LO_FREQUENCY	34	Y	double64	Hz	y		Front end tuner frequency. N/A for direct sampling front ends
FM_SNR	66		float32	dB			Estimated SNR in FM; equal to channel SNR when moments squelch is disabled
FREQ_OFFSET	59		float32	Hz			Estimated signal frequency error. Only when PLL is enabled in linear modes
GAIN	68	Y	float32	dB			Demodulator gain (constant for FM, variable for linear)
GAINSTEP	81		uint		Y		Front end analog gain, arbitrary units, hardware specific
GPS_TIME	3		uint	ns	Y		Nanoseconds since GPS epoch of 6 January 1980 00:00:00 UTC. Generated by front end, passed through 'radio'
HEADROOM	63	Y	float32	dBFS		<= 0	Target channel output audio level, block average
HIGH_EDGE	40	Y	float32	Hz		LOW_EDGE < HIGH_EDGE < +Fs/2	Upper edge of post-mixer filter
IF_GAIN	27	Y	uint	dB	Y	0-255	Relative gain of baseband analog amplifier in tuner just ahead of A/D converter. Hardware dependent, not used by all front ends
IF_POWER	45		float32	dBFS	Y		A/D output level
INDEPENDENT_SIDEHAND	50	Y	boolean				LSB in left channel, USB in right channel - currently unimplemented
INPUT_SAMPLES	13		uint				Count of input data samples
INPUT_SAMPRATE	10		uint	Hz			Sample rate of RTP input data stream
IQ_IMBALANCE	30		float32		Y		Relative gain of I and Q channels (only direct conversion front ends). 1 = no error
IQ_PHASE	31		float32	radians	Y	-pi/2 to +pi/2	Relative phase error of I & Q channels. 0 = no error
KAISER_BETA	41	Y	float32			>= 0	Kaiser β factor for downconverter filter window design
LNA_GAIN	25	Y	uint	dB	Y	0-255	Relative gain of analog input to receiver. Hardware dependent, not used by all front ends
LOCK	78		boolean				Will ignore frequency tuning commands. Not yet implemented
LOW_EDGE	39	Y	float32	Hz		-Fs/2 < LOW_EDGE < HIGH_EDGE	Lower edge of post-mixer filter
MINPACKET	72	Y	uint	frames		0-10	Minimum number of receiver frames in an output IP packet, unless packet is already at MTU
MIXER_GAIN	26	Y	uint	dB	Y	0-255	Relative gain of mixer in analog receiver/downconverter. Hardware dependent, not used by all front ends
NOISE_BW	15		float32	bins		>0	Relative noise bandwidth of each FFT bin in spectral display, depends on WINDOW_TYPE and SPECTRUM_SHAPE. 1 for rectangular
NOISE_DENSITY	47		float32	dBmJ			Estimated noise spectral power density, N0, in and near downconverter channel
OPUS_APPLICATION	112	Y	uint			2048 (VOIP), 2049 (AUDIO), 2051 (Low Delay)	The application type parameter to the Opus encoder.
OPUS_BANDWIDTH	113	Y	uint	Hz		1101 (4 kHz), 1102 (6 kHz), 1103 (8 kHz), 1104 (12 kHz) 1105 (20 kHz)	Input bandwidth to be considered important by Opus encoder
OPUS_BITRATE	71	Y	uint	bits/sec			Target bitrate of Opus-compressed audio. 0 = auto
OPUS_DTX	111	Y	boolean				enable discontinuous transmission by the Opus encoder
OPUS_FEC	114	Y	uint	percent		0-100	Frame loss percentage the Opus encoder should use when in FEC mode
OUTPUT_CHANNELS	49	Y	uint			1 - 2	mono (=1) stereo (=2); for front ends, 1 channel = real, 2 channels = complex (IQ)
OUTPUT_DATA_DEST_SOCKET	17	Y	socket				Destination (multicast) IP address and port of output data stream
OUTPUT_DATA_PACKETS	22		uint	count			Count of RTP output data packets
OUTPUT_DATA_SOURCE_SOCKET	16		socket		Y		Source IP and port of output RTP data stream
OUTPUT_ENCODING	107	Y	uint				1=S16LE, 2=S16BE, 3=Opus, 4=F32BE, 5=AX25, 6=F16BE, 7=Opus with application=voip
OUTPUT_ERRORS	23		uint	count			Count of send errors on channel output packet stream
OUTPUT_LEVEL	69		float32	dBFS			Output level, frame average
OUTPUT_METADATA_PACKETS	21		uint	count	y		Count of metadata packets sent
OUTPUT_SAMPLES	70		uint	samples			Output sample count
OUTPUT_SAMPRATE	20	Y	uint	Hz		>0	Sample rate of RTP output data stream
OUTPUT_SSRC	18	Y	uint			32 bits; 0 and ffffffff reserved	RTP stream ID of output stream
OUTPUT_TTL	19		uint	hops	Y	0 <= ttl <= 255	IP Time-to_live (hop count limit) of output data stream (not metadata, which can be different)
PEAK_DEVIATION	60		float32	Hz		>= 0	Peak deviation (FM demodulators only)
PL_DEVIATION	89		float32	Hz		>= 0	Measured deviation of PL tone (FM demodulator only; tone squelch enabled)
PL_TONE	61		float32	Hz		>= 0	PL tone squelch frequency (FM demodulator only); 0 = no tone
PLL_BW	55	Y	float32	Hz		>0	Noise bandwidth of PLL loop filter
PLL_ENABLE	51	Y	boolean				Enable 0 Hz carrier tracking & squelch (Linear mode); PLL demodulation (FM)
PLL_LOCK	52		boolean				Indicate whether PLL is in lock (controlled by squelch threshold settings)
PLL_PHASE	54		float32	radians?		0 - 2 pi	Relative phase of PLL numerically controlled oscillator
PLL_SNR	58		float32	dB			Phase lock loop signal-to-noise ratio; = 10log10(1/2/Q^2 - 1), in-phase to quadrature power ratio
PLL_SQUARE	53	Y	boolean				Square feedback to PLL; use for DSB AM and BPSK. Implies PLL_ENABLE
PLL_WRAPS	109		signed int	rotations			Count of complete 360 degree (2 pi radian) rotations of PLL
PRESET	85	Y	string			defined in presets.conf	Set demodulator mode - configured by modes.conf on 'radio'
RADIO_FREQUENCY	33	Y	double64	Hz	Y	>0 for real inputs. 0 reserved to mean "idle channel"	RF tuning frequency that comes out of the downconverter at 0 Hz. I.e, "carrier frequency"
RESOLUTION_BW	93	Y	float32	Hz		positive	Width of each bin in spectrum data
RF_AGC	99	Y	boolean		Y		Enable front end RF automatic gain control
RF_ATTEN	97	Y	float32	dB	Y	<= 0	Front end attenuation, hardware dependent (not present on all front ends). Setting turns off RF_AGC
RF_GAIN	98	Y	float32	dB	Y		Front end gain; hardware dependent. Setting turns off RF_AGC
RF_LEVEL_CAL	110		float32	dBm	Y		Input power that gives 0 dBFS from A/D converter with all RF gain and atten = 0
RTP_PT	105		uint			0-127	Real Time Protocol Payload Type for current output stream
RTP_TIMESNAP	8		uint			32 bits	Snapshot of RTP 32-bit timestamp field
SAMPLES_SINCE_OVER	108		uint		Y		Input A/D samples since last overrange event
SECOND_LO_FREQUENCY	35		double64	Hz			Digital down converter frequency = -(RADIO_FREQUENCY+DOPPLER_FREQUENCY-FIRST_LO_FREQUENCY). Negative of IF frequency. <0 for direct sampling front ends. May be >0 or <0 for direct conversion (I/Q) front ends. >0 for tuners with high-side injection, <0 for tuners with low-side injection.
SETOPTS	6	Y	uint			32 bits	1-bits turn on specified option (32 max)
SHIFT_FREQUENCY	36	Y	double64	Hz			Post-downconversion shift frequency, used primarily for CW. With IF filter centered, <0 shifts the LSB up in output frequency. >0 shifts the USB up in output frequency
SNR_SQUELCH	57	Y	boolean				Y=enable SNR squelch in all modes. N=enable moments squelch (in FM mode only)
SPECTRUM_AVG	12	Y	uint	frames		>= 1	Number of consecutive periodograms (power spectra) to be averaged in each response.
SPECTRUM_BASE	11	Y	float32	dB			Used to interpret BIN_BYTE_DATA
SPECTRUM_FFT_N	76		uint	bins		> 0	Number of bins in analysis FFTs. Calculated from BIN_COUNT and RESOLUTION_BW
SPECTRUM_OVERLAP	116	Y	float32			0-1	If SPECTRUM_AVG > 1, sets overlap of averaged FFT windows. 0=no overlap, 1=complete overlap
SPECTRUM_SHAPE	91	Y	float32			>= 0	Shape factor for spectrum analysis window (kaiser β, gaussian α)
SPECTRUM_STEP	115	Y	float32	dB		non-negative	step size of 8-bit bins in v2 spectrum data
SQUELCH_CLOSE	84	Y	float32	dB			Squelch closing threshold (FM, synchronous AM) - must be less than or equal than SQUELCH_OPEN
SQUELCH_OPEN	83	Y	float32	dB			Squelch opening threshold (FM, synchronous AM)
STATUS_DEST_SOCKET	5		socket		y		Source IP address and port number for control and status
STATUS_INTERVAL	106	Y	uint	frames	y		Number of frames between periodic channel status beacons on output stream
THRESH_EXTEND	90	Y	boolean				Enable/disable threshold extension (FM only)
TP1	79		float32				General purpose test point #1
TP2	80		float32				General purpose test point #2
WINDOW_TYPE	14	Y	uint			0=Kaiser 1=Rectangular(none) 2=Blackman 3=Exact Blackman 4=Gaussian 5=Hann 6=Hamming 7=Blackman-Harris	Window applied to FFT input data in spectrum analysis. Kaiser and Gaussian require SPECTRUM_SHAPE parameter, β and α respectively.

Encoding notes:

“uint” is a variable length unsigned integer sent in big-endian order (MSB first).

“boolean” is an integer with value 0 (false/off) or 1 (true/on).

“float32” is a 32-bit IEEE-754 floating point number set as 4 bytes, sign byte (MSB) first.

double64 is a 64-bit IEEE-754 floating point number sent as 8 bytes, sign byte (MSB) first.

“socket” is an IPv4 or IPv6 address in big-endian order followed by a 2-byte port number in big-endian order.

A uint, float32 or double64 with value 0 may be sent as a zero-length value field.

A string is a UTF-8 character string that may or may not be null terminated because the length is explicitly specified in the prefix.

A “settable” parameter is one that can be changed with this protocol sent as a command packet. Some other parameters may be set through the configuration file. The rest are output-only, eg., counters and calculated values.

Packets are encoded as follows: 1 byte to indicate status (0) or command (1), followed by a series of TLV-encoded items. The first byte is the numerical type as shown in this table. The second byte encodes the length; values 0-127 indicate that the value that follows uses 0-127 bytes. A length of zero encodes the value 0 as either an integer, float32, double, or boolean (false). A length of 128-256 indicate a value >127 bytes long; subtract 128 from this field to obtain the length in bytes of the actual length field that will immediately follow. If > 1 byte long, the length bytes are interpreted in big endian order (MSB first).

The BIN_DATA (96) option contains a list of float32s. The number of floats is the length divided by 4.

Items marked “Global” are common to all channels; the rest are unique to a channel.