ka9q-radio command/status protocol element types - Phil Karn, KA9Q - Jan 2026

Type	num	Settable?	Encoding	Units	Global	Valid range/values
AD_BITS_PER_SAMPLE	82		uint	bits	Y	
AD_OVER	104		uint		Y	
AGC_ENABLE	62	Y	boolean			
AGC_HANGTIME	64	Y	float32	sec		>= 0
AGC_RECOVERY_RATE	65	Y	float32	dB/sec		>= 0
AGC_THRESHOLD	67	Y	float32	dBFS		<= 0
BASEBAND_POWER	46		float32	dB		
BIN_BYTE_DATA	9		bytes			Limited by IP packet size
BIN_COUNT	94	Y	uint	bins		> 0
BIN_DATA	96		float32 vector			
CALIBRATE	24	Y	double64		Y	
CLEAROPTS	7	Y	uint			32 bits
CMD_CNT	2		uint		Y	
COHERENT_BIN_SPACING	92		float32		y	
COMMAND_TAG	1	Y	uint		Y	
CONVERTER_OFFSET	88		float32	Hz	Y	
CROSSOVER	95	Y	float32			positive
DC_I_OFFSET	28		float32		Y	
DC_Q_OFFSET	29		float32		Y	
DEEMPH_GAIN	87		float32	dB		
DEEMPH_TC	86		float32	s		non-negative
DEMOD_TYPE	48	Y	uint	enum		0 - 4
DESCRIPTION	4		string		Y	free-form UTF-8
DIRECT_CONVERSION	32		boolean		Y	
DOPPLER_FREQUENCY	37	Y	double64	Hz		
DOPPLER_FREQUENCY_RATE	38	Y	double64	Hz/sec		
ENVELOPE	56	Y	boolean			
EOL	0					
FE_HIGH_EDGE	101		float32	Hz	Y	> FE_LOW_EDGE
FE_ISREAL	102		boolean		Y	
FE_LOW_EDGE	100		float32	Hz	Y	<FE_HIGH_EDGE
FILTER_BLOCKSIZE	42		uint	samples	Y	>0
FILTER_DROPS	77		uint	blocks		
FILTER_FIR_LENGTH	43		uint	samples	Y	>0
FILTER2	44	Y	uint	frames		0-10
FILTER2_BLOCKSIZE	73		uint	samples		>0
FILTER2_FIR_LENGTH	74		uint	samples		>0
FILTER2_KAISER_BETA	75	Y	float32			>=0
FIRST_LO_FREQUENCY	34	Y	double64	Hz	y	
FM_SNR	66		float32	dB		

Type	num	Settable?	Encoding	Units	Global	Valid range/values
FREQ_OFFSET	59		float32	Hz		
GAIN	68	Y	float32	dB		
GAINSTEP	81		uint		Y	
GPS_TIME	3		uint	ns	Y	
HEADROOM	63	Y	float32	dBFS		<= 0
HIGH_EDGE	40	Y	float32	Hz		LOW_EDGE < HIGH_EDGE < +Fs/2
IF_GAIN	27	Y	uint	dB	Y	0-255
IF_POWER	45		float32	dBFS	Y	
INDEPENDENT_SIDEBAND	50	Y	boolean			
INPUT_SAMPLES	13		uint			
INPUT_SAMPRATE	10		uint	Hz		
IQ_IMBALANCE	30		float32		Y	
IQ_PHASE	31		float32	radians	Y	-pi/2 to +pi/2
KAISER_BETA	41	Y	float32			>= 0
LNA_GAIN	25	Y	uint	dB	Y	0-255
LOCK	78		boolean			
LOW_EDGE	39	Y	float32	Hz		-Fs/2 < LOW_EDGE < HIGH_EDGE
MINPACKET	72	Y	uint	frames		0-10
MIXER_GAIN	26	Y	uint	dB	Y	0-255
NOISE_BW	15		float32	bins		>0
NOISE_DENSITY	47		float32	dBmJ		
OPUS_APPLICATION	112	Y	uint			2048 (VOIP), 2049 (AUDIO), 2051 (Low Delay)
OPUS_BANDWIDTH	113	Y	uint	Hz		1101 (4 kHz), 1102 (6 kHz), 1103 (8 kHz), 1104 (12 kHz) 1105 (20 kH)
OPUS_BITRATE	71	Y	uint	bits/sec		
OPUS_DTX	111	Y	boolean			
OPUS_FEC	114	Y	uint	percent		0-100
OUTPUT_CHANNELS	49	Y	uint			1 - 2
OUTPUT_DATA_DEST_SOCKET	17	Y	socket			
OUTPUT_DATA_PACKETS	22		uint	count		
OUTPUT_DATA_SOURCE_SOCKET	16		socket		Y	
OUTPUT_ENCODING	107	Y	uint			1-7
OUTPUT_ERRORS	23		uint	count		
OUTPUT_LEVEL	69		float32	dBFS		
OUTPUT_METADATA_PACKETS	21		uint	count	y	
OUTPUT_SAMPLES	70		uint	samples		
OUTPUT_SAMPRATE	20	Y	uint	Hz		>0
OUTPUT_SSRC	18	Y	uint			32 bits; 0 and ffffffff reserved
OUTPUT_TTL	19		uint	hops	Y	0 <= ttl <= 255
PEAK_DEVIATION	60		float32	Hz		>= 0
PL_DEVIATION	89		float32	Hz		>= 0
PL_TONE	61		float32	Hz		>= 0

Type	num	Settable?	Encoding	Units	Global	Valid range/values
PLL_BW	55	Y	float32	Hz		>0
PLL_ENABLE	51	Y	boolean			
PLL_LOCK	52		boolean			
PLL_PHASE	54		float32	radians?		0 - 2 pi
PLL_SNR	58		float32	dB		
PLL_SQUARE	53	Y	boolean			
PLL_WRAPS	109		signed int	rotations		
PRESET	85	Y	string			defined in presets.conf
RADIO_FREQUENCY	33	Y	double64	Hz	Y	>0 for real inputs. 0 reserved to mean “idle channel”
RESOLUTION_BW	93	Y	float32	Hz		positive
RF_AGC	99	Y	boolean		Y	
RF_ATTEN	97	Y	float32	dB	Y	<= 0
RF_GAIN	98	Y	float32	dB	Y	
RF_LEVEL_CAL	110		float32	dBm	Y	
RTP_PT	105		uint			0-127
RTP_TIMESNAP	8		uint			32 bits
SAMPLES_SINCE_OVER	108		uint		Y	
SECOND_LO_FREQUENCY	35		double64	Hz		
SETOPTS	6	Y	uint			32 bits
SHIFT_FREQUENCY	36	Y	double64	Hz		
SNR_SQUELCH	57	Y	boolean			
SPECTRUM_AVG	12	Y	uint	frames		>= 1
SPECTRUM_BASE	11	Y	float32	dB		
SPECTRUM_FFT_N	76		uint	bins		> 0
SPECTRUM_OVERLAP	116	Y	float32			0-1
SPECTRUM_SHAPE	91	Y	float32			>= 0
SPECTRUM_STEP	115	Y	float32	dB		non-negative
SQUELCH_CLOSE	84	Y	float32	dB		
SQUELCH_OPEN	83	Y	float32	dB		
STATUS_DEST_SOCKET	5		socket		y	
STATUS_INTERVAL	106	Y	uint	frames	y	
THRESH_EXTEND	90	Y	boolean			
TP1	79		float32			
TP2	80		float32			
WINDOW_TYPE	14	Y	uint			0=Kaiser 1=Rectangular(none) 2=Blackman 3=Exact Blackman 4=Gaussian 5=Hann 6=Hamming 7=Blackman-Harris
unused	103					

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.

Type	Meaning & Use
AD_BITS_PER_SAMPLE	Width of A/D input word
AD_OVER	Count of A/D overrange events
AGC_ENABLE	Automatic gain control (Linear demod only)
AGC_HANGTIME	Time delay before automatic gain increase on lowered signal (linear demod only)
AGC_RECOVERY_RATE	Gain increase rate on lowered signal (linear demod only)
AGC_THRESHOLD	Target demodulator output level on noise only
BASEBAND_POWER	Signal power at channel downconverter filter output, relative to unity
BIN_BYTE_DATA	Vector representing spectrum data, 1 bin/byte, unsigned, with meaning SPECTRUM_BASE + SPECTRUM_STEP * x dB
BIN_COUNT	Number of FFT bins wanted in spectrum data
BIN_DATA	Vector of 4-byte float32s with linear spectrum data, spectrum mode only. Order: DC…max positive freq, max neg freq…-1
CALIBRATE	Frequency calibration factor for tuner reference and A/D sample clock. Actual freq = nominal * (1 + CALIBRATE), ie, 0 means “on frequency”
CLEAROPTS	1-bits turn off specified option (32 max)
CMD_CNT	Server count of received commands
COHERENT_BIN_SPACING	deprecated; can be calculated from INPUT_SAMPRATE, FILTER_BLOCKSIZE, FILTER_FIR_LENGTH
COMMAND_TAG	generated by controller, echoed by server to confirm command
CONVERTER_OFFSET	Frequency offset of external frequency converter. Not yet implemented
CROSSOVER	Value of RESOLUTION_BW above which the wideband spectrum analyzer is used.
DC_I_OFFSET	DC offset of I-channel A/D converter (only direct conversion front ends)
DC_Q_OFFSET	DC offset of Q-channel A/D converter (only direct conversion front ends)
DEEMPH_GAIN	Static gain correction when de-emphasis used to maintain subjectively equal loudness
DEEMPH_TC	Deemphasis time constant (0 = off), FM only
DEMOD_TYPE	Demodulator type, enum: 0 = linear; 1 = FM/PM; 2 = Wideband FM with stereo demodulator; 3 = spectrum; 4 = spectrum v2
DESCRIPTION	description of front end (antenna, etc). Generated by front end, passed through ‘radio’. Need not be null terminated.
DIRECT_CONVERSION	Front end uses direct conversion with DC spike and 1/f noise that should be avoided
DOPPLER_FREQUENCY	Doppler tuning offset (untested)
DOPPLER_FREQUENCY_RATE	Rate of change of Doppler tuning effort (untested). Limited to 1 bin/frame time
ENVELOPE	Use envelope detector in linear demodulator
EOL	End of option list. No length or value field
FE_HIGH_EDGE	Upper edge of A/D converter input frequency band (negative for tuners with high side injection)
FE_ISREAL	Y: front end uses a single A/D converter; N: front end uses dual (I/Q) A/D converters
FE_LOW_EDGE	Low edge of A/D converter input frequency band (negative for tuners with high side injection)
FILTER_BLOCKSIZE	Input samples per downconverter FFT processing block
FILTER_DROPS	Number of frame drops by digital downconverter
FILTER_FIR_LENGTH	Overlap samples per downconverter FFT processing block. Sets maximum impulse duration of downconverter channel filter
FILTER2	Size of secondary filter input in units of downconverter frames (eg, 20 ms). 0=off
FILTER2_BLOCKSIZE	Input samples per Filter2 block
FILTER2_FIR_LENGTH	Impulse response length in secondary filter (filter2)
FILTER2_KAISER_BETA	Kaiser β factor for secondary filter (filter2) window design
FIRST_LO_FREQUENCY	Front end tuner frequency. N/A for direct sampling front ends
FM_SNR	Estimated SNR in FM; equal to channel SNR when moments squelch is disabled

Type	Meaning & Use
FREQ_OFFSET	Estimated signal frequency error. Only when PLL is enabled in linear modes
GAIN	Demodulator gain (constant for FM, variable for linear)
GAINSTEP	Front end analog gain, arbitrary units, hardware specific
GPS_TIME	Nanoseconds since GPS epoch of 6 January 1980 00:00:00 UTC. Generated by front end, passed through ‘radio’
HEADROOM	Target channel output audio level, block average
HIGH_EDGE	Upper edge of post-mixer filter
IF_GAIN	Relative gain of baseband analog amplifier in tuner just ahead of A/D converter. Hardware dependent, not used by all front ends
IF_POWER	A/D output level
INDEPENDENT_SIDEBAND	LSB in left channel, USB in right channel - currently unimplemented
INPUT_SAMPLES	Count of input data samples
INPUT_SAMPRATE	Sample rate of RTP input data stream
IQ_IMBALANCE	Relative gain of I and Q channels (only direct conversion front ends). 1 = no error
IQ_PHASE	Relative phase error of I & Q channels. 0 = no error
KAISER_BETA	Kaiser β factor for downconverter filter window design
LNA_GAIN	Relative gain of analog input to receiver. Hardware dependent, not used by all front ends
LOCK	Will ignore frequency tuning commands. Not yet implemented
LOW_EDGE	Lower edge of post-mixer filter
MINPACKET	Minimum number of receiver frames in an output IP packet, unless packet is already at MTU
MIXER_GAIN	Relative gain of mixer in analog receiver/downconverter. Hardware dependent, not used by all front ends
NOISE_BW	Relative noise bandwidth of each FFT bin in spectral display, depends on WINDOW_TYPE and SPECTRUM_SHAPE. 1 for rectangular
NOISE_DENSITY	Estimated noise spectral power density, N0, in and near downconverter channel
OPUS_APPLICATION	The application type parameter to the Opus encoder.
OPUS_BANDWIDTH	Input bandwidth to be considered important by Opus encoder
OPUS_BITRATE	Target bitrate of Opus-compressed audio. 0 = auto
OPUS_DTX	enable discontinuous transmission by the Opus encoder
OPUS_FEC	Frame loss percentage the Opus encoder should use when in FEC mode
OUTPUT_CHANNELS	mono (=1) stereo (=2); for front ends, 1 channel = real, 2 channels = complex (IQ)
OUTPUT_DATA_DEST_SOCKET	Destination (multicast) IP address and port of output data stream
OUTPUT_DATA_PACKETS	Count of RTP output data packets
OUTPUT_DATA_SOURCE_SOCKET	Source IP and port of output RTP data stream
OUTPUT_ENCODING	1=S16LE, 2=S16BE, 3=Opus, 4=F32BE, 5=AX25, 6=F16BE, 7=Opus with application=voip
OUTPUT_ERRORS	Count of send errors on channel output packet stream
OUTPUT_LEVEL	Output level, frame average
OUTPUT_METADATA_PACKETS	Count of metadata packets sent
OUTPUT_SAMPLES	Output sample count
OUTPUT_SAMPRATE	Sample rate of RTP output data stream
OUTPUT_SSRC	RTP stream ID of output stream
OUTPUT_TTL	IP Time-to_live (hop count limit) of output data stream (not metadata, which can be different)
PEAK_DEVIATION	Peak deviation (FM demodulators only)
PL_DEVIATION	Measured deviation of PL tone (FM demodulator only, tone squelch enabled)
PL_TONE	PL tone squelch frequency (FM demodulator only); 0 = no tone

Type	Meaning & Use
PLL_BW	Noise bandwidth of PLL loop filter
PLL_ENABLE	Enable 0 Hz carrier tracking & squelch (Linear mode); PLL demodulation (FM)
PLL_LOCK	Indicate whether PLL is in lock (controlled by squelch threshold settings)
PLL_PHASE	Relative phase of PLL numerically controlled oscillator
PLL_SNR	Phase lock loop signal-to-noise ratio; = 10log10(I^2/Q^2 - 1), in-phase to quadrature power ratio
PLL_SQUARE	Square feedback to PLL; use for DSB AM and BPSK. Implies PLL_ENABLE
PLL_WRAPS	Count of complete 360 degree (2 pi radian) rotations of PLL
PRESET	Set demodulator mode - configured by modes.conf on 'radio'
RADIO_FREQUENCY	RF tuning frequency that comes out of the downconverter at 0 Hz. I.e, "carrier frequency"
RESOLUTION_BW	Width of each bin in spectrum data
RF_AGC	Enable front end RF automatic gain control
RF_ATTEN	Front end attenuation, hardware dependent (not present on all front ends). Setting turns off RF_AGC
RF_GAIN	Front end gain; hardware dependent. Setting turns off RF_AGC
RF_LEVEL_CAL	Input power that gives 0 dBFS from A/D converter with all RF gain and atten = 0
RTP_PT	Real Time Protocol Payload Type for current output stream
RTP_TIMESNAP	Snapshot of RTP 32-bit timestamp field
SAMPLES_SINCE_OVER	Input A/D samples since last overrange event
SECOND_LO_FREQUENCY	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	1-bits turn on specified option (32 max)
SHIFT_FREQUENCY	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	Y=enable SNR squelch in all modes. N=enable moments squelch (in FM mode only)
SPECTRUM_AVG	Number of consecutive periodograms (power spectra) to be averaged in each response.
SPECTRUM_BASE	Used to interpret BIN_BYTE_DATA
SPECTRUM_FFT_N	Number of bins in analysis FFTs. Calculated from BIN_COUNT and RESOLUTION_BW
SPECTRUM_OVERLAP	If SPECTRUM_AVG > 1, sets overlap of averaged FFT windows. 0=no overlap, 1=complete overlap
SPECTRUM_SHAPE	Shape factor for spectrum analysis window (kaiser β, gaussian α)
SPECTRUM_STEP	step size of 8-bit bins in v2 spectrum data
SQUELCH_CLOSE	Squelch closing threshold (FM, synchronous AM) - must be less than or equal than SQUELCH_OPEN
SQUELCH_OPEN	Squelch opening threshold (FM, synchronous AM)
STATUS_DEST_SOCKET	Source IP address and port number for control and status
STATUS_INTERVAL	Number of frames between periodic channel status beacons on output stream
THRESH_EXTEND	Enable/disable threshold extension (FM only)
TP1	General purpose test point #1
TP2	General purpose test point #2
WINDOW_TYPE	Window applied to FFT input data in spectrum analysis. Kaiser and Gaussian require SPECTRUM_SHAPE parameter, β and α respectively.
unused	