EEProm Setup Record For "Barix CIP enabled SIP Client"

Application	Parameter	Offset [dec]	Dynamic Name	Length [byte]	Default Value	Short Description
All	Volume	500	B500	1	10	Linear volume in 5% steps for the range 0-100%, values 0-20
All	AD-Gain	501	B501	1	2	A/D amplifier gain in 1.5dB steps starting from -3dB (value 0) to 19.5dB (value 15)
All	Mic-Gain	502	B502	1	0	Microphone gain in 1.5dB steps starting from 21dB (value 0) to 43.5dB (value 15)
All	Input Mode	510	B510	1	0x82	Input Mode: 0x01 = Line In 0x02 = Mic In 0x80 = Mono
All	Encoding Type	511	B511	1	6	0 = MPEG2/22.05kHz 1 = MPEG1/44.1kHz 2 = MPEG2/24kHz 3 = MPEG1/48kHz 4 = MPEG2/16kHz 5 = MPEG1/32kHz 6 = uLaw/24kHz 7 = uLaw/8kHz 8 = aLaw/24kHz 9 = aLaw/8kHz 10 = PCM/24kHz 11 = PCM/8kHz 12 = G722/16 kHz
All	Encoding Quality	512	B512	1	0	MPEG encoding quality, 0 (lowest) to 7(highest)
SIP	Call on input 0	640	S640	31		Predefined sip extension/URI to call on input 0.
SIP	Call on input 1	672	S672	31		Predefined sip extension/URI to call on input 1.
SIP	Peer to peer	900	B900	1	disabled	Enable/disable peer to peer mode.
SIP	Sip port	901	W901	2	5060	Sip protocol port.
SIP	RTP port	903	W903	2	5004	RTP protocol port.
SIP	Username	905	S905	31		Sip protocol username.
SIP	Password	938	S938	31		Sip protocol password.

Application	Parameter	Offset [dec]	Dynamic Name	Length [byte]	Default Value	Short Description
SIP	Call on level	955	S955	31		Call on level sip extension/URI.
SIP	PBX or remote peer	1020	S1020	31		PBX or remote peer ip/host.
SIP	NAT keep alive	1053	B1053	1	disabled	Enable/disable sending of NAT keep alive.
SIP	Debug mode	1054	B1054	1	disabled	Enable/disable debug syslog messages.
SIP	Phone pickup mode	1055	B1055	1	1	0 - autoanswer. 1 - autopick up after timeout. 2 - autohang up after timeout. 3 - not callable.
SIP	Pick up / hang off after	1056	B1056	1	20	Number of seconds for auto pick-up / hang off.
SIP	Open door code	1057	S1057	8		Door open DTMF sequence.
SIP	Open door relay timeout	1066	B1066	1	1	Door open timeout.
SIP	API interface UDP control port	1067	W1067	2	0	API UDP control port.
SIP	API interface TCP control port	1069	W1069	2	0	API TCP control port.
SIP	Serial Control Support	1084	B1084	1	0	Sets what serial COM port 1 is used for: 0 – No use 1 - Serial Control Interface
SIP/CIP/RAVA	Relay No to Enable at Call Answer/Ring	1087	B1087	1	0	Set here what the relay should be used for: 0 – used for door open 1 – enable relay on call answer 2 – enable the relay on call ring Default: used for door open
SIP	Current SIP Profile	1088	B1088	1	0	SIP client profile. Can select between the following profiles: 0 - SIP Phone 1 - SIP Door Station 2 - SIP Paging Station 3 - SIP Paging Gateway 4 - SIP Monitoring Point
SIP	Close On Level	1089	B1089	1	0	Close the call on level timeout: 0 - disabled

Application	Parameter	Offset [dec]	Dynamic Name	Length [byte]	Default Value	Short Description
						5,10,15,20,25,30 - seconds
SIP	Auto Hang up Time	1090	B1090	1	0	Auto hang up outgoing call in door station mode on timeout: 0 - disabled 5,10,15,20,25,30 - seconds
SIP	Talk Mode	1091	B1091	1	0	0 – Full Duplex (FDX), 1- Half Duplex (HDX
SIP	Output trigger Level	1093	W1093	2	1000	Defines the output audio level which will trigger the SIP client in Listen Mode Values : from 0 to 32767
SIP	Trigger Level Timeout	1095	W1095	2	500	Defines the timeout in milliseconds, after which if no audio is detected from the incoming RTP stream, the SIP Client should switch back to "talk" mode. This setting is ignored if the door station is configured in FDX mode
SIP	DNS renewal	1116	B1116	1	0	Enable (1) or disable (0) the DNS renewal mechanism in case of REGISTER failure
SIP	Registration fail timeout	1117	B1117	1	10	Sets the timeout in seconds after which the registration will be considered as "failed" in case of no reply from the SIP server. In this case the SIP client will resolve the DNS address of the SIP server, and try again
SIP	Stream timeout	1118	W1118	2	0	The time in minutes to close the active call if there is no incoming audio stream detected. "0" is disabled (default)
SIP	Default REGISTER Time	1120	W1120	2	1800	The value that the SIP client suggests to the SIP server when sending the REGISTER request.
SIP	AEC	1122	B1122	1	0	Enable (1), or disable (0) the Acoustic Echo Cancellation (AEC)
SIP	Beep on call answer	1123	B1123	1	0	Enable , or disable the playback of beep tone when the call is answered bit0 – 0 (disable beep on outgoing call answer), 1 – enable bit1 – 0 (disable beep on incoming call answer), 1 – enable
SIP/CIP/RAVA	CIP port	1124	W1124	2	41794	CIP control port
SIP/CIP/RAVA	SIP Groups	1126	S1126	50		Used in CIP SIP requests to get the SIP groups in which the device is a member of
SIP/CIP/RAVA	SIP Display ID	1176	S1176	24		Used in CIP SIP requests to set the ID to be displayed on the remote party when ringing.