

EEProm Setup Record For “Barix 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 timeout	639	B639	1	0	0 – disabled 1-255 – timeout for closing the call in minutes
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	Call on input 2	704	S704	31		Predefined sip extension/URI to call on input 2 (Annunicom 1000).
SIP	Call on input 3	736	S736	31		Predefined sip extension/URI to call on input 3 (Annunicom 1000).
SIP	Call on input 4	768	S768	31		Predefined sip extension/URI to call on input 4 (Annunicom 1000).
SIP	Call on input 5	800	S800	31		Predefined sip extension/URI to call on input 5 (Annunicom 1000).

Application	Parameter	Offset [dec]	Dynamic Name	Length [byte]	Default Value	Short Description
SIP	Call on input 6	832	S832	31		Predefined sip extension/URI to call on input 6 (Annunicom 1000).
SIP	Call on input 7	864	S864	31		Predefined sip extension/URI to call on input 7 (Annunicom 1000).
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	32		Sip protocol username.
SIP	Password	938	S938	32		Sip protocol password.
SIP	Call on level	970	S970	31		Call on level sip extension/URI.
SIP	SIP Display ID	1002	S1002	16		Description to be displayed on the remote end when calling
SIP	PBX or remote peer	1020	S1020	32		PBX or remote peer ip/host. Use as PBX 1 server in case of using the client in SIP mode
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	Rebroadcast IP address	1071	B1071, B1072, B1073, B1074	4	0	Set the audio rebroadcast destination IP address.
SIP	Rebroadcast port	1075	W1075	2	0	Set the audio rebroadcast destination port. A value of 0 disable the audio rebroadcasting.

Application	Parameter	Offset [dec]	Dynamic Name	Length [byte]	Default Value	Short Description
SIP	BGM ip address	1077	B1077, B1078, B1079, B1080	4	0	Set the BGM listening address.
SIP	BGM port	1081	w1081	2	0	Background music listening port.
SIP	BGM volume	1083	B1083	1	15	Set the background music volume.
SIP	Serial Control Support	1084	B1084	1	1	Sets what serial COM port 1 is used for: 0 - X8 Extensions 1 - Serial Control Interface 2 - Serial Gateway, UDP 3 - Serial Gateway, TCP, passive 4 - Serial Gateway, TCP, active
SIP	BGM buffer	1085	W1085	2	600	Set the background music buffer in ms.
SIP	Relay No to Enable at Call Answer/Ring	1087	B1087	1	0	Set here the relay number (1-8) that you would like to have automatically switched on. Bit7: 0 – enable the relay on call answer; 1 – on incoming call ring Bits0-4: Relay number to activate. If 0 then the feature is disabled Default: Disabled
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 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	AI Phone Support	1092	B1092	1	0	0 – Disabled, 1 - Enabled
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

Application	Parameter	Offset [dec]	Dynamic Name	Length [byte]	Default Value	Short Description
SIP	Notification audio port	1097	W1097	2	0	The port number to which the device should listen for notification messages. Default: 0 (disabled)
SIP	Notification port IP address	1099	B1099, B1100 B1101, B1102	4	0	Notification Audio Source IP address.
SIP	Notification Volume	1103	B1103	1	16	Notification Audio Volume. Default: 80%
SIP	Notification Audio Buffer	1104	W1104	2	600	Notification audio RTP buffer. Default: 600 ms
SIP	Notification Relay	1107	B1107	1	0	Sets if to use, or not to use the INBOUND CALLS “Relay No to Enable at Call Answer” (B1087). Values: 0 – do not use, 1 – switch on the INBOUND CALLS relay while the notification message is active.
SIP	Serial GW Address	1108	B1108,B1109, B1110, B1111	4	0.0.0.0	Sets the TCP/UDP address for the serial Gateway
SIP	UDP GW port	1112	W1112	2	12302	Sets the UDP port number to send packets to (in case of active UDP gateway), or the port number to listen (in case of passive UDP gateway)
SIP	TCP GW port	1114	W1114	2	12301	Sets the remote TCP port number to establish a connection to (in case of active TCP gateway), or the TCP port number to listen for incoming connections (in case of passive UDP gateway)
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	Auth ID	1124	S1124	32	0	Authentication ID (if used). When configured, it is used instead of the SIP Username (S905)
SIP	PBX 2	1157	S1157	32		PBX 2 server in case of using the client in SIP mode
SIP	REFER enable	1190	B1190	1	0	Enable (1), or disable (0) the support for the SIP REFER method (used for Blind Call Transfer feature)

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SIP	OffHook/Onhook call	1191	B1191	1	0	Enable (1), or disable (0) the offhook/onhook call mode. When enabled, pressing the button makes the call, releasing it cancels/closes it Note: It works only on Annunicom 60 in SIP Phone profile (mode 0)
SIP	SSRC type	1192	B1192	1	0	Defines SSRC type. MAC Based (0), Random (1). When “MAC based” is set, the SSRC is calculated from the last 5 HEX letters of the MAC. When set to “Random”, generates new SSRC for every new call.
SIP + X8	call on X8 input 0	1200	S1200	31		Predefined sip extension/URI to call on X8 input 0.
SIP + X8	call on X8 input 1	1232	S1232	31		Predefined sip extension/URI to call on X8 input 1.
SIP + X8	call on X8 input 2	1264	S1264	31		Predefined sip extension/URI to call on X8 input 2.
SIP + X8	call on X8 input 3	1296	S1296	31		Predefined sip extension/URI to call on X8 input 3.
SIP + X8	call on X8 input 4	1328	S1328	31		Predefined sip extension/URI to call on X8 input 4.
SIP + X8	call on X8 input 5	1360	S1360	31		Predefined sip extension/URI to call on X8 input 5.
SIP + X8	call on X8 input 6	1392	S1392	31		Predefined sip extension/URI to call on X8 input 6.
SIP + X8	call on X8 input 7	1424	S1424	31		Predefined sip extension/URI to call on X8 input 7.