<Appendix>

WPU-7800G Administrator Manual

Configuration Entry (v 1.0.0)

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1. Overview

1. This is a document explained all configuration entry, default and valid value so that administrator can understand them for better environment of WPU-7800G.
2. This configuration entry should be used on WPU-7800G provided by UniData only and should not be used on other product.
   i. Web configuration tool, PC application (PhoneConfigGenerator), APS system
   ii. Please refer to administrator's manual for usage of each tool.
3. Please find the administrator's manual for details
4. This document is subject to change if entry is added.

2. Configuration entry

Section	Entry name	Default value	Comment	Valid Value
SYSTEM	Language	1	Displaying the UI supported language	English(1)/Italian(39)/Dutch(31)/French(33)/German(49)/Portuguese(351)/Spanish(34)/
	Admin_Password	000000	Setting for admin password	6 String
	Country_Tone_Type	1	Setting for countries tone	U.S(1)/ South Africa(27)/ Greece(30)/ Netherlands(31)/ Belgium(32)/ Spain(34)/ Italy(39)/ Switzerland(41)/ Austria(43)/ United Kingdom(44)/Denmark(45)/ Sweden(46)/ Germany(49)/ Brasil(55)/ Japan(81)/ Korea(82)/China(86)/ Hong Kong(852)
	Use_DNS_SRV	1	Whether or not to use SRV Record in DNS Query	1: Use, In time of failure, try A Record again. 0: Attempt only with A Record.
RTP_RTCP	Use_RTCP	TRUE	if value is TRUE, it use RTCP (Real Time Transport Protocol)	0 :disable, 1:enable
	RTP_Port_Min	9000	Setting for minimum RTP(Real Time Transport Protocol) port value Notice: If the value is more than PTP_port_Max, Function could be limited.	1024 ~ 65535
	RTP_Port_Max	9020	Setting for Maximum RTP(Real Time Transport Protocol) port value Notice: If value is less than PTP_port_Max, Function could be limited	1024 ~ 65535

	RTCP_Report_Interval	5000	Setting for RTCP interval if Value=0, it means there is no interval. Unit: sec	0 ~ 65535
	RTCP_CNAME	"WPU-7800"	Character String to enter CNAME of RTCP Message	0 ~ 24 string
	Last_RTP_Received_Timeout	0	The call could be dropped, when value is smaller than last received RTP packet's value 0 means disable. Notice: If Hold or session time are enable this value will ignored.	0 ~ 65535
WEB SERVER	Enable_Web_Server	1	To use WEB Server Function or not (PC-SYNC)	0 :disable, 1:enable
	User_Password		Set to PC-Sync Password	0~ 12
TIME	Enable_NTP	1	Time setting by using NTP or not.	0 :disable, 1:enable
	Date_Format	0	Time display format	0 : YYYY/MM/DD Sun 1 : YY/MM/DD Sun 2 : MM/DD/YYYY Sun 3 : MM/DD/YY Sun 4 : DD Jan,YY (Sun)
	Time_Zone_Index	"+09:00"	Setting GMT offset +09:00 or 9:00 or -03:00 or -3:00 etc. Colon(:) divides Hour and Minute, so must not omit colon(:)	
	Enable_Daylight_Saving_Tim e	1	Set to Summer time value	0 :disable, 1:enable
	DST_Start_Month	3	DST Start Month	1 ~ 12 mouth
	DST_Start_Day	8	DST Start Day	1~31 day
	DST_End_Month	11	DST End Month	1 ~ 12 mouth
	DST_End_Day	1	DST End Day	1~31 day
	NTP_Refresh_Interval	7200	Setting for NTP(Netowork Time Protocol) refresh invervalUnit: Sec	60 ~ 3153600
	NTP_Server1	203.248.240.103	Setting for primary Time Server	IP address or FQDN
	NTP_Server2	203.254.163.74	Setting for secondary Time Server	IP address or FQDN
SIP	Local_Port	5060	SIP has a standard Port, which is usually 5060. Notice: The SIP port is mandatory if there is no port for SIP, the call will be failed	5000 ~ 40000
	T1	500	SIP T1 Timer	100~: 65535
	T2	4000	SIP T2 Timer	100~: 65535
	T4	5000	SIP T4 Timer	100~: 65535

	Timer B	0	SIP Timer B (INVITE Transaction timeout timer)	500~: 65535
	Timer F	0	SIP Timer F (non-INVITE Transaction timerout timer)	500~: 65535
	Use_User_Agent	0	Add User-Agent Header in SIP Message or not	0 : not use, 1 : use
	User_Agent_Name	"WPU-7800"	String in User-Agent Header	1~60 string
	Use_Version_On_User_Agent	0	Add version in User-Agent Header	0: not use, 1: use
	Use_Vendor_ID_On_User_Agent	0	Add servername in User-Agent Header in User-Agent Header	0: not use, 1: use
	Use_MAC_On_User_Agent	0	Add mac address in User-Agent Header in User-Agent Header	0: not use, 1: use
	Max_Forwards	70	Max-Forwards Header Value of SIP Message	0~999
	Retry_Hold_On_491	1	Whether or not to retry in case of receiving 491 for response of hold request	0 : noop, 1 : retry
	Caller_ID_Mode	0	Setting Caller ID	0 : Default, 1 : Privacy ,3325 : RFC3325
	ICT_Transaction_Max_Count	10	ICT Transaction Max value	0~100
	Use_rport	0	Whether or not to use rport	0 : not use, 1 : use
	Options_Expire	20	rfc3581 : in case received and rport in via Header of response different from local ip, send options periodically	0 ~86400
	Request_REFER_Timeout	200	Request REFER Method Timeout. unit : ms	200 ~ 32000 0:disable
	Wait_REFER_Response_Timeout	800	Wait REFER Method Response timeout unit : ms	500 ~ 32000 0:disable
	180_Retransmission_Interval	0	180 response retransmission interval, 0 is off this function. (unit: sec)	1~3600 0:disable
	Invite_Expire	180	the value of Expires header on INVITE (refer to RFC3261 13.3.1), 0 is off this function. (unit: sec)	10 ~86400 0:disable
	Reuse_Auth_Header_Within_Dialog	0	reuse authorization header within same dialog	0 :disable, 1:enable
	Reuse_Auth_Header_On_Register	0	reuse authorization header on REGISTER	0 :disable, 1:enable
	Register_Contact_Change	0	In changing IP in Registerinbg, not send old_contact in on.	0 :disable, 1:enable
	Use_Remove_All_Contact	1	In expire=0, the problem not recognizing contact="*", select one of "*"/ IP for Contact Value.	0 :disable, 1:enable
	Use_Random_Contact	0	Use userinfo Portion of contact header with random number instead of alias	0 :disable, 1:enable
	Retry_To_Redirect_Contact	1	try to redirect when phone is received 3xx response	0 :disable, 1:enable

	Auth_Retry_Count	1	how many times retry request again when server response 401/407.	1~10
	Retry_Register_After_Auth_Fail	0	retry regsiter after authentication fail	0 :disable, 1:enable
	Ingore_183_Without_SDP	0	Disregard when 183 has come without SDP.	0 :disable, 1:enable
	Illegal_UserName_Check_At_Request_URI	0	When username and IP bring different information equest-URI in time of incomming call, 404 response occurs.	0 :disable, 1:enable
	Remove_Register_On_Start	0	First send Unregister when starting first after Booting.	0 :disable, 1:enable
	Check_Escape_Of_Call_Number	0	escape character in call number check.(ex # ->%23)	0 :disable, 1:enable
	Transaction_Delete_Before_L3HANDOVER	0	Transaction's delete before L3HANDOVER	0 :disable, 1:enable
SDP	Use_Increase_session_id	0	Whether or not to increase owner's session id in case the content of sdp changes	0 : not increase, 1 : increase
	Use_Increase_version	1	Whether or not to increase owner's version in case the content of sdp changes	0 : not increase, 1 : increase
	Modified_Session_Detection	0	This entry changes SDP version, SDE session ID by sever when changed session. At this time, server cannot check change if values are not equal. For that reason, check by using this entry.	0 : not detect 1 : SDP Version detect 2 : SDP Session ID detect
	Session_Name	"A_conversation"	session name on sdp	0~30 string
USER_ACCOUNT	Displayname	NULL	Setting for LCD display when phone is idle. If value is null, telephone number will be displayed This entry is added in version 2.4.0	30 String
	Phone_Number	NULL	Setting for phone number Notice: If value is null, the call will be failed	32 String
	User_ID	NULL	Setting for SIP user ID Notice: Unless this value do not match with user ID in SIP server , Regiter will be failed	32 String
	User_Password	NULL	Setting for SIP user Password Notice: Unless this value do not match with user password in SIP server , Register will be failed	32 String
	URL_Scheme	0	Setting SIP URL Scheme	0 : sip, 1 : tel
SERVER_SETTINGS	1st_Proxy	NULL	Setting for 1st proxy server	IP address or FQDN

	2nd_Proxy	NULL	Setting for 2nd proxy server	IP address or FQDN
	Domain_Realm	NULL	Setting for proxy domain	FQDN
	Register_Expire	3600	Setting for Register expire intervals in SIP	60 ~ 86400
	Register_Retry_Backoff_Interval	"60,120,240,480,960"	If not registered, it periodically retries by this timer	0~50 csv format (unit : sec)
REDUNDANCY	Mode	0	Set REDUNDANCY mode	0 : registration status oriented mode 1 : master/slave mode
	Request_Timeout	4000	If no response from one of servers when using REDUNDANCY, set timeout for initial request on order to move to next server.	500~ 32000 0:disable
	Use_Fixed_Primary_Server	1	Whether or not to fix Primary Server	0 : not fix, 1 : fix
	Use_DNS_Additional_Records	1	Use dns additional records part for redundancy	0 :disable, 1:enable
BASIC_CALL	Use_Headset_Button	0	Use Headset Button	0 :disable, 1:enable
	Busy_Tone_Count	10	busytone repeated count when call end	1~60
	Transfer_Busy_Tone_Count	10	busytone repeated count when transfer failed	1~60
	Call_Waiting_Tone_Count	1	call waiting tone count	-1 : infinte 0 : disable 1 ~ 10 : count
	Reject_Hold_Request_On_Hold	1	forbidden hold request during on hold or on transfer	1: forbidden, 0: permit
	Reject_Hold_Request_On_2Calls	0	forbidden hold request during on having 2 calls	1: forbidden, 0: permit
	Block_Request_Hold_On_Holded	0	Whether Hold Reqest of Held Telephone or not	0 : Hold Permited 1:Hold not Permited
	Use_Call_Waiting	TRUE	This value is true, call waiting function is also possbile.	TRUE or FALSE
	Session_Expire	1800	setting for Session Expire interval	90 ~ 3600 0:disable
	Ringing_Timeout	180	When ringing, this timer is over, rejects a call and becomes the call in absence	6~3600
	Update_RBT	0	Update RBT by every 18x response	0: standard, 1: always update, 2: new dialog
	Dial_Sending_Timeout	30000	INVITE expire (ms)	500~32000 0:disable
	Check_Alias_In_Call_List	1	Check if a certain alias is in call list whose line is busy, and if it is, not make a new call	0 : not check, 1 : check
	Remove_DASH_On_Alias	1	Whether or not to delete '-' in the midway of alias	0 : remain, 1 : remove

	Use_Silent_Packet_On_Mute	0	Whether or not to send silent RTP packet in Mute to remote	0 : not use, 1 : use
	Echo_Canceller	1	use echo canceller	0 :disable, 1:enable
	Reject_Invite_During_BT	1	reject invite during BT	1 : Enable, 0: Disable
	Response_For_Decline_Incoming_Call	603	response for decline incoming call 488 or 603	488 or 603
	Call_Disconnect_TimeOut	0	Set to timer when the network is disconnected during Call busy or Ringing	500~ 32000 0:disable
	Use_INVITE_Filtering	1	Use Filtering the incomming invite	0 :disable, 1:enable
HOLD	Mode	2543	Setting for hod mode Notice: Value should be eqaul with SIP proxy value. If the value is different, call hold can be impossible.	1: Inband hold (hold tone play to RTP without SIP signaling) 2543 : c=0.0.0.0 (RFC2543) 3264 :Use sendonly, recvonly, inactive attribute (RFC3264) 0 : 2543 + 3264 mixed default : 2543
	Use_Local_Hold_Tone	1	Whether or not to play Hold tone at the terminal in its own way	0 : not use, 1 : standard(c=0 received, local tone play) 2: all mode(RFC3264, also local tone play))
	RTP_Hold_Multiframe	40	unit : ms (not recommend to use 20ms in time of using g729a)	20~100
	RTP_Hold_Play_Wait_Time	500	In time of melody hold, play after a certain time.	0~10000 0:disable
	Enable_Hold_With_Inactive	FALSE	If the value is True, SDP value in the hold invite will send a=inactive This value only can be used when Hold mode is 3264 or mixed. Notice: If the value do not match with SIP Proxy, normal Hold could be impossible.	TRUE or FALSE
	Enable_RTP_Hold	1	No RTP send and set hold tone when receiving "sendonly"'	1 : Enable, 0: Disable
MWI	Use_MWI	TRUE	If the value is True, it means MWI is enabled. False means diabled.	TRUE or FALSE
	Use_Subscribe	TRUE	If the value is True, it means SIP SUBSCRIBE Method is registered in MWI Server.False means diabled.	TRUE or FALSE
	Subscribe_Server	NULL	Setting for Subscriber server Unless the value is NULL, it sends SIP subscribe to the registerd server. Notice: If ths value of Use_Subscribe is TRUE and Subscribe_Server is NULL, it sends SIP SUBSCRIBE to the 1st_Proxy.	IP address or FQDN

	Subscribe_Expire	3600	Setting for MWI SUBSCRIBE expire Unit: Sec	180 ~ 65535
	VMS_Alias	NULL	Setting for MWI account Unless the value is NULL, It will connect MWI and set it to the Request-URI of "INVITE" or "to Header" Notice: If Message-account is set in SIP NOFITY Method from MWI server, VMS_Alia will be renewed.	extention number or SIP Request-URI
TRANSFER	Use_Transfer_Target_Hold	1	Whether or not to send Hold Request to Transfer Target before Transferor sends REFER	0 : not use, 1 : use
	Use_Consultation_Transfer	1	Whether or not to use the function of Consultation Transfer x	0 : not use, 1 : use
	Use_Blind_Transfer	1	Whether or not to use the function of Blind Transfer (Unattended Transfer)	0 : not use, 1: use
	Use_100_NOTIFY	1	Whether or not to send Transferee 100Trying NOTIFY to Transferor	0 : not use, 1: use
	Use_Attended_Transfer_On_Incomming_Call	1	When WPU-7700 has to calls and current talking call is "incomming call", you can disable attended transfer by setting this entry to '0'	0 : not use, 1: use
	Use_Contact_For_Refer_To	1	Whether to make Refer-To header refering to Contact Header or to SIP URL of Transfer Target	1: Refer to Contact, 0: Refer to SIP URL
	Use_Attended_Transfer_On_Call_Switch	1	In case of 2 calls, if 1st call is busy, not to connect to attended transfer.	1 : enable transfer, 0 : disable transfer
	Check_BYE_After_Refer	0	In case of receiving BYE before receiving NOTIFY(200) after transferer sends refer, mark transfer fail	0 : disable, 1 : enable
	Enable_Enterprise_Blind_Transfer	0	Set to minimalize the UI for transfer	0 : disable, 1 : enable
FORWARD	Use_Forward	0	Whether or not to use forwarding feature.	1 : Use Forward, 0 : Nonuse Forward
	Mode	0	Setting forward mode when Use_Forward is enable	0 : Disable 1 : All 2 : Busy 3 : No Answer
	Phone_Number	NULL	set the forward number	0~30
	No_Answer_Timeout	10	Set time of no response when No Answer mode. Forwarding, if this time is over.	2~3600
	Disable_Call_Waiting_On_Busy	0	In case Forward is Busy mode, when a call is connected, Forward is OK as well.	0 : disable, 1 : enable

INITIAL BUSY MODE	Sendkey_Block_Timeout	1	Set to input prohibit timeout value of <Send Key> after answer Incoming call	0~5
ROAMING	Roaming_Disable	0	Whether or not to use roaming feature	1: disable, 0 : enable
	Try_Beacon_Signal_Level	-77	Setting for Roaming Roaming will be started when Beacon signal is lower than default value.	-103 ~ -64
	Try_Rx_Signal_Level	-77	When Tx Error occurs over value Count, attempt Roaming	-103 ~ -64
	Roaming_Disable_On_Calling	0	Set to disable roaming when the line is engaged	0 : disable, 1 : enable
TOS_WMM	Precedence_0	0	set priority of ToS precedence 0	0~7
	Precedence_1	6	set priority of ToS precedence 1	0~7
	Precedence_2	6	set priority of ToS precedence 2	0~7
	Precedence_3	6	set priority of ToS precedence 3	0~7
	Precedence_4	6	set priority of ToS precedence 4	0~7
	Precedence_5	6	set priority of ToS precedence 5	0~7
	Precedence_6	6	set priority of ToS precedence 6	0~7
	Precedence_7	6	set priority of ToS precedence 7	0~7
NETWORK (1,2,3,4)	Enable	TRUE	True=using Network 1. False=Network1 disabled. Notice: Network 1 cannot be deleted or edit. This only can be edited by Auto Provisioning.	TRUE or FALSE
	SIP_OutBound_Proxy	NULL	Settiing for SIP outbound Proxy Notice: SIP Outbound proxy should be set to every individual Network profile.	IP address or FQDN
	SSID	voip	Setting for SSID in Network 1 When the phone searching AP, the SSID will be appear. "voip" is a default value	32 String
	Enable_DHCP	TRUE	True=Using DHCP Notice: False=Put the IP mannually	TRUE or FALSE
	Address	0.0.0.0	Setting for Phone IP address Notice: If Enable_DHCP value is True, this value cannot be applied.	IP address
	Netmask	255.255.255.0	Setting for Netmaks Notice: If Enable_DHCP value is True, this value cannot be applied.	IP address

	Gateway	0.0.0.0	Setting for Gateway Notice: If Enable_DHCP value is True, this value cannot be applied.	IP address
	DNS1	0.0.0.0	Setting for 1st DNS Notice: If Enable_DHCP value is True, this value cannot be applied.	IP address
	DNS2	0.0.0.0	Setting for 2st DNS Notice: If Enable_DHCP value is True, this value cannot be applied.	IP address
	Security	2	Selecting AP authentication security Notice: Phone and AP should be use same type of authentication security.	0 : none, 1 : WEP, 2 : WPA-PSK, 3 : WPA2-PSK, 4 : WPA-EAP, 5 : WPA2-EAP
	WEP_Bits	0	Setting for WEP KEY bitNotice:Phone and AP should be use same WEP key	0 : 64 bit, 1 : 128bit
	Default_WEP_Key	1	Setting for default WEB Key index Notice: Phone and AP should be use same WEP key	1 ~ 4
	WEP_Key1	NULL	Setting 1st WEB Key index Notice: Phone and AP should be use same WEP key	87 String
	WEP_Key2	NULL	Setting 2nd WEB Key index Notice: Phone and AP should be use same WEP key	87 String
	WEP_Key3	NULL	Setting 3rd WEB Key index Notice: Phone and AP should be use same WEP key	87 String
	WEP_Key4	NULL	Setting 4th WEB Key index Notice: Phone and AP should be use same WEP key	87 String
	Post_Authentication_Mode	0	Setting for 802.1x Authentication Notice:Phone and AP should be use same ID, Password and Authentication	0 : None, 1 : WEB, 2 : 8021X-MD5, 3 : 8021X-TLS, 4 : 8021X-PEAP, 5 : 8021X-TTLS
	8021X_Name	NULL	Setting ID for 802.1x Authentication Notice:Phone and AP should be use same ID	50 String
	8021X_Password	NULL	Setting password for 802.1x Authentication Notice:Phone and AP should be use same password	50 String
	WPA_PSK_PassPhrase	NULL	Setting PassPhrase for WPA PSK Key Notice: Security should be selected as WPA PSK. Do not Set WPA_PSK Pass Phrase, if you use PSK Key Should be set Only WPA_PSK_Key field.	63 String
	WPA_PSK_Key	NULL	Setting for WPA-PSK keyNotice:Security should be selected WPA PSK.Do not set WPA-PSK_Key, if you use WPA-PSK PassPhrase, Notice:A pass-phrase is a sequence of between 8 and 63 ASCII-encoded characters but WPU Phone UI Menu support only 18 characters.	63 String

	Proactive_Key_Caching	FALSE	This entry can be used to enable proactive key caching which is alos known as opportunistic PMKSA caching for WPA2. Enable by setting this to TRUE.	TRUE or FALSE
	DiffServ_Signal	46	Setting DSCP(Differentiated Service Code Point) for DiffServ Signal	0 ~ 63
	DiffServ_Media	46	Setting DSCP(Differentiated Service Code Point) for DiffServ Media	0 ~ 63
	WMM	TRUE	Setting for WMM True=WMM enable	TRUE or FALSE
	Jitter_Buffer_Size	60	Setting for Jitter buffer size units: ms	20 ~ 200
	Bell_ID	1	Setting for bell sound Simple Ball / Sadness Ball / Glad Ball / Happy Ball Notice:should be set the value by hexadecimal	Simple Bell : 0x00000001 ~ 0x00000006 Sadmess Bell : 0x00000101 ~ 0x00000106 Glad Bell : 0x00000201 ~ 0x00000210 Happy Bell : 0x00000301 ~ 0x00000307
	Bell_Volume	6	Setting for bell volums Notice:0=No sound	0 ~ 12
DTMF	Mode	0	DTMF Transmission System	0 : In-Audio 1 : SIP-INFO 2 : rfc2833 3: In-Audio+SIP-INFO
	Duration	100	Duration value of SIP INFO, ms	0~1000
	RFC2833_Volume	10	volume value of 2833 Packet	0~63
	RFC2833_Payload_Type	101	Payload type for RFC2833	96~127
	RFC2833 Packet Count	10	DTMF Packet count	3~20
	RFC2833 End Count	9	DTMF End packet count	3~20
	Enable_Auto_DTMF_Mode	0	Decision method of DTMF transmission system	0 : Transmit DTMF with entry "Mode" of Section[DTMF] unconditionally. 1 : If telephone-event in sdp, set RFC2833, if not, set inband. (can define sip info in sdp by the vendor. In case there is such payload type in Sdp offer, can select sdp answer according to the preferred order.)
CALLER_ID	Use_Caller_ID_OnOff	1	Use or not use the function of Caller ID On/Off	1: Use, 0: Not Use
	Enable_Caller_ID	1	Whether or not send CID after Enable	0 : disable, 1 : enable
	Anonymous_Displayname	"Anonymous"	String to be used in display name of From in time of INVITE Request with caller id off	0~30 string

	Use_Update_Caller_ID	0	In case cid changes in the midway of call, update cid or not	0 : not use, 1 : use
	Hide_Displayname	0	Whether or not mark display name in time of Caller_ID display	0 : show, 1 : hide
	Update_Caller_ID_After_Transfer	1	transferee and transfer target display real talking remote party after transfer (remain call log also)	0:off 1:on 2:update display ony
SMS	Message_Server	NULL	Message server address	0~50 string
	Message_Content_Type	0	SIP MESSAGE Content-Type	0 : text/plan 1 : text/html
	Use_Register	0	register to IM server	0 : disable, 1 : enable
	Register_Expire	3600	IM expire	60~ 86400
	SMS_Prefix	NULL	Set to SMS Prefix	0~12
	Receive_Display	0	Set to indication for SMS on the display when receive SMS	0 : disable, 1 : enable
	Inform_Interval	1	SMS indication interval	0~2
	Auto_Save	0	Set to Auto save SMS	0 : disable, 1 : enable
	Spam_Num0~9	NULL	Set to SPAM number	0~32 string
	Spam_Str0~9	NULL	Set to SPAM String	0~32 string
DHCP OPTION	Use_Option_2	0	Use DHCP option 2 (Time offset)	0 : disable, 1 : enable
	Use_Option_42	0	Use DHCP option 42 (NTP handout)	0 : disable, 1 : enable
	Use_Option_66	0	Use DHCP option 66 (TFT server name)	0 : disable, 1 : enable
PROVISION	Use_Provision	0	Use provision or not use	0 : Disable, 1 : Enable
	Request_Mode	2	Select Request Protocol	0 : HTTP 1 : HTTPS 2 : TFTP
	Firmware_URL	NULL	IP or Domain of Firmware(Do not input protocol)	IP address or FQDN
	Firmware_Version	NULL	version of firmware	0~30 string
	Firmware_Name	NULL	Name of firmware	0~30 string
	Enable_HTTP_Keep_Alive	1	KeepAlive ON/OFF	0 : Disable, 1 : Enable
	Phonebook_Name	NULL	Setting for phonebook file such as csv type Enter the phone book file name. Load that file on the TFTP server that the WPU-7700 pulls its configuration from and it will update when you turn the phone off and on. This entry is added from version 2.4.0	128 String
UNICODE	Use_Unicode_On_SMS	0	Whether or not to use unicode(utf-8) with content of SMS	0 : local charset

				1 : utf-8
	Use_Unicode_Displayname	0	Whether or not to use unicode(utf-8) with content of Displayname	0 : local charset 1 : utf-9
CALL BLOCK	Phone_Number1~3	NULL	Set to Call block number	0~32 string
	SIP Response Code	NULL	Set to SIP Response code block	0~ 999