## List of timers, unnamed bits, bytes, words and long words, that cannot be accessed via configuration windows

Date 08-16-2012, firmware release 3.13

| timers, Unnamed bits, bytes, words or long words | | description | default |
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| timer | | Call from ISDN reaction delay<br><br>In very few cases PBXs have problems if the reaction to a SETUP is too fast. A delay can be defined at this point. | *0.10 sec.* |
| timer | | Waiting time for first digit from ISDN<br><br>The AS55X waits this time for the first digit of ISDN suffix dialing. After this timer has expired, the call-setup attempt from ISDN domain will be aborted. | *15 sec.* |
| timer | | Waiting time for reconnect from ISDN<br><br>When a call-setup to an ISDN number is made and the ISDN phone is ringing, the AS55X will wait this time for an answer from the ISDN user. | *120 sec.* |
| timer | | Time limit of callback attempt to ISDN<br><br>If a callback from GSM domain to ISDN domain is in progress, the AS55X will wait this time for an answer from the ISDN user. | *60 sec.* |
| timer | | Call from GSM reaction delay<br><br>From the sight of the GSM network, the GSM channel of the AS55X is a mobile phone. Sometimes GSM networks do not behave properly, if the reaction of a call to a mobile phone is too fast. With this setting, the reaction delay can be adjusted. | *1.00 sec.* |
| timer | | Waiting time for CLIP from GSM<br><br>Some GSM networks do not send the CLIP of the calling subscriber immediately with the ring command and there is a small delay. The AS55X is waiting this time for the CLIP from the GSM network before processing the call. | *2 sec.* |
| timer | | Waiting time for first DTMF digit from GSM<br><br>The AS55X waits this time for the first digit of DTMF suffix dialing from the mobile phone. This timer is only active, if no fix subscriber delay for this call or this channel is defined. After this timer has expired, the call-setup attempt from GSM domain will be aborted. | *15 sec.* |
| timer | | Waiting time for next DTMF digit from GSM<br><br>The AS55X waits this time for the next digit of DTMF suffix dialing from the mobile phone. After this timer has expired, the call-setup attempt from GSM domain will be aborted. | *15 sec.* |
| timer | | Time limit of call to fix subscriber<br><br>If a call to a fix subscriber is in progress, the AS55X is waiting this time for an answer of the called user. After this timer has expired, the GSM caller can select another ISDN subscriber by DTMF dialing.<br><br>Notice: If the caller is enabled for DTMF dialing after an undelayed fix subscriber call has been made, there will be traffic charges to the mobile phone during suffix dialing. | *60 sec.* |
| timer | | Waiting time for connect from GSM<br><br>When a call-setup to a GSM number is made and the GSM phone is ringing, the AS55X will wait this maximum time for an answer from the GSM user. After this timer has expired, the call-setup attempt will be aborted. | *120 sec.* |
| timer | | Blocking time of GSM channel after call termination<br><br>During this time, the GSM channel of an AS55X does not accept a new call-setup from the ISDN domain after the previous call has ended.<br><br>In some cases, especially with high GSM network load phases, it can be necessary to have a short timeout between ending of a call and starting the next call. | *1.0 sec.* |

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| timer | | Delay before start of GME active Task<br><br>If the DTMF tones for entering the GME code are quite long, with this timer the start of the new active task can be delayed in order to avoid recognizing the rest of the tone by this new task. | *0.50 sec.* |
| timer | | Get GSM cell information repetition time in idle mode<br><br>For troubleshooting in case of GSM network problems, the information about the GSM cells can be helpful. Such cases can be the search of the best antenna position for investigation of GSM call setup problems.<br><br>The distances of reading the cell informations in idle mode and in active mode (call present) can be set separately.<br><br>The informations can be displayed via the trace functions. | *0 sec.* |
| timer | | Get GSM cell information repetition time in active mode<br><br>For troubleshooting in case of GSM network problems, the information about the GSM cells can be helpful. Such cases can be the search of the best antenna position for investigation of GSM call setup problems.<br><br>The distances of reading the cell informations in idle mode and in active mode (call present) can be set separately.<br><br>The informations can be displayed via the trace functions. | *0 sec.* |
| timer | | SIP OPTIONS request repetition period<br><br>The presence detection of the peers in SIP peer mode and of the clients in SIP server mode is made with a SIP OPTIONS request. This function can additionally work as NAT keep alive.<br><br>With this setting, the distance of two OPTIONS requests can be specified. | *10.00 sec.* |
| timer | | SIP registration repetition<br><br>In SIP client mode, the AS55X must register at its server for a period of time and then the registration must be repeated. If you choose a short period, the LAN load is high. A too long period may cause problems when working via a DSL modem if you are getting a new public IP address. The value of 300 seconds is a good compromise. | *300 sec.* |
| bit | 7 | Unnamed bit 7, do not process national emergency numbers<br><br>If set, national emergency numbers are not extra processed. This bit has no affect to the GSM emergency number 112. | *set* |
| bit | 9 | Unnamed bit 9, trace GSM engine startup sequence<br><br>If set, the startup sequence of the GSM engines will be traced. This bit can be used to decrease trace load if modules are missing SIMs. | *set* |
| bit | 10 | Unnamed bit 10, use GSM network nationality<br><br>If set, the GSM network nationality will be used for CLIP converting from GSM if CLIP does not appear in international order. If this bit is not set, the SIM nationality will be used. | *not set* |
| bit | 11 | Unnamed bit 11, allow callback to GSM before callback via call setup list<br><br>If both callback via call setup list and callback to the GSM participant apply to one call from GSM, the callback to the GSM participant will be executed first if this bit is set. If it is not set, the callback GSM will be suppressed in this case. | *not set* |
| bit | 14 | Unnamed bit 14: Load audio parameters unconditional<br><br>If set, the audio parameters for the GSM module will be loaded unconditional with every module startup. | *not set* |
| bit | 15 | Unnamed bit 15: Never load audio parameters<br><br>If set, the audio parameters for the GSM module will never be loaded. | *not set* |

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| bit | 16 | Unnamed bit 16, GME code suppression<br><br>If set, the DTMF tones for the GME code in direction to wired interface will be suppressed. | *set* |
| bit | 17 | Unnamed bit 17, write information element channel identification unconditional<br><br>If set, this ISDN information element will be included in every call setup message. Otherwise, it will be included according to ETSI rules. | *not set* |
| bit | 26 | Unnamed bit 26, use address place holder in via header<br><br>If set, the actual IP address and port number are not included in SIP via header. They are replaced by place holders. This bit setting might be necessary, if a NAT router is placed between AS55X and PBX (or SIP provider). | *not set* |
| bit | 28 | Unnamed bit 28, allow active SIP call without audio<br><br>If set, an available audio channel is not necessary for successfully setting up of a SIP call. This setting is helpful, if the PBX defines the audio channel not till the call is active. | *not set* |
| bit | 29 | Unnamed bit 29, redirect RTP stream to received address / port<br><br>If this bit is set and if the source address / port of the received RTP stream are not the same as the values in the session description protocol (SDP), the AS55X redirects the sending of the RTP stream to the source of the received one. This setting can be necessary, if a NAT router is placed between AS55X and PBX (or SIP provider). | *set* |
| bit | 31 | Unnamed bit 31, SimSwitch count minutes instead of seconds<br><br>Helpful for SimSwitch demonstration or evaluation. If set, the second counter are running 60 times faster. | *not set* |
| bit | 32 | Unnamed bit 32, keep remote logical address header for moved<br><br>This bit must be set for SIP call forwarding or redirection with 'moved'. | *not set* |
| bit | 34 | Unnamed bit 34, detect '+CMUX:' as sent short message<br><br>Test setting: For evaluating of short message counting by SimSwitch, the response of the module to the command 'AT+CMUX=0' can be handled as sent short message. | *not set* |
| bit | 35 | Unnamed bit 35, suppress DHCP request as connection check<br><br>If the AS55X work as DHCP client, it generates a cyclic DHCP request as connection check to the network. If this causes errors in the network, the cyclic request can be suppressed with this bit. | *not set* |
| bit | 36 | Unnamed bit 36, deactivate local host queue<br><br>Test setting: LAN packet, sent to the own IP address will not be handled internally and they will be sent via the LAN. | *not set* |
| bit | 40 | Unnamed bit 40, accept any OPTIONS response<br><br>For the detection of the connection to the SIP peers, OPTIONS requests are sent to these addresses. The connection normally will be detected if the peer responses with OK. If this bit is set, the connection is counted as given, if the peer sends back any SIP response. | *not set* |
| bit | 46 | Unnamed bit 46, do not allow SIP IP and port redirection<br><br>There is an information in a SIP request, where to send the response. In some cases, especially in contact with NAT, this value is not set correctly. If this unnamed bit is set, the AS55X does always send the response to that IP address and port, it receives the request from. | *not set* |
| bit | 47 | Unnamed bit 47, use local username for SIP registration<br><br>If an authentication identity is entered for a SIP account, it will be used for authentication for registration and call-setups. If this bit is set, for registration the username is used as authentication identity. | *not set* |
| bit | 48 | Unnamed bit 48, get public IP address from register response | *not set* |

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| bit | 49 | Unnamed bit 49, get public port from register response<br><br>In a NAT context, it might be possible to use the public IP address and port in the SIP protocol as some SIP providers do only accept call-setups with the correct values. With this bit the fetching of these informations from the register response message can be controlled. | *not set* |
| bit | 50 | Unnamed bit 50, abort SIP connection with TCP socket close<br><br>Normally, for a SIP connection it is not really necessary that the correspondent TCP connection is up for the whole time. If this bit is set, the SIP connection will be aborted, if the TCP port is closed. | *not set* |
| bit | 52 | Unnamed bit 52, allow registrar IP as proxy identity<br><br>In SIP client mode normally call setups are only accepted if they correspond to that entity, the AS55X is registered at. A random number in the registration contact is used to recognize the entity, but some providers do not handle this contact properly. If this bit is set, it is sufficient, if the call setup arrives from the IP address of the registrar. | *not set* |
| bit | 55 | FAX speed restriction<br>If this bit is set, FAX speed is limited to 4800bps. This setting only makes sense, if due to bad line quality, the training procedures often gets very long. | *not set* |
| bit | 56 | no FAX function restriction<br>Normally enhanced FAX modes like: error correction, colour transmission etc. are disabled as this is strongly recommended with FAX over GSM. These modes can be enabled by setting this bit. | *not set* |
| bit | 59 | do not store local parameters<br>For faster synchronisation of both FAX devices during incoming calls, the parameters of the local FAX device will be stored in the WLL. If this bit is set, there is no storing and the parameters will newly be read from the local FAX device with every call. | *not set* |
| bit | 60 | do not store remote parameters<br>For faster synchronisation of both FAX devices during outgoing calls, the parameters of some remote FAX devices will be stored in the WLL. If this bit is set, there is no storing and the parameters will newly be read from the remote FAX device with every call. | *not set* |
| bit | 65 | add a leading ‘0’ zero to called party number<br>Sometimes PBXs need an additional leading '0' if the destination of the call is not in the PBX domain (number of the wired network). Depending on the detection of PBX internal and external numbers, this digit will be added will be added to the number dialled into the PBX. | *not set* |
| byte | 12 | Unnamed byte 12, traffic direction<br><br>These bytes specify the allowed call setup directions:<br>1: Calls from PBX to GSM are allowed<br>2: Calls from GSM to PBX are allowed<br>3: Calls in both directions are allowed | *3* |
| byte | 19 | Unnamed byte 19, overrun type of callback number<br><br>If in case of a callback via call setup list, the PBX does not expect the callback number with that type of the CLIP of the first call to GSM, the number type can be changed with this byte.<br>0: Unknown type<br>1: Subscriber type<br>2: National prefix and digits<br>3: National type and digits<br>4: International prefix, country code and digits<br>5: International type, country code and digits | *255 (decimal), no overrun* |

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| byte | 29 | Unnamed byte 29, DTMF processing<br><br>This byte holds the configured DTMF processing modes.<br>1: Detect restart sequence (*0) during call setup from GSM to PBX<br>2: Detect restart sequence (*0) from GSM during active call<br>4: Convert SIP telephone event to DTMF to GSM<br>8: Convert DTMF from GSM to telephone event to SIP<br>Code the byte as sum of the single modes. The telephone event (RFC2833/4733) conversion modes are also settable with DTMF processing in SIP access configuration. | *3* |
| byte | 56 | Unnamed byte 56, call waiting off trials<br><br>This byte specifies the trials of switching call waiting off with the startup procedure of the module. If the GSM network does not accept this command, the startup of the module gets faster if this value is set to 0. | *1* |
| byte | 59 | Unnamed byte 59, GME timeout passive line<br><br>This byte specifies the time limit, a GME line is allowed to be in passive state. This timer shall avoid that forgotten lines block resources. | *120 (decimal)* |
| byte | 63 | Unnamed byte 63, type of CLIP to ISDN for AUX interface<br><br>The same as 'Type of CLIP to ISDN', settable with general ISDN access, but not for the main interface.<br>0: Unknown type<br>1: Subscriber type<br>2: National prefix and digits<br>3: National type and digits<br>4: International prefix, country code and digits<br>5: International type, country code and digits | *2* |
| byte | 66 | Unnamed byte 66, SIP server registration mode<br><br>1: SIP clients have to register explicit with REGISTER request (normal mode)<br>2: SIP clients can also register implicit with an INVITE request | *1* |
| byte | 69 | Unnamed byte 69, IP trace mode<br><br>1: Trace received frame<br>2: Trace sent frame<br>4: Trace received fragments<br>Code this byte as sum of the single modes. The trace of received and sent frames can also be switched on or off by configuration window 'advanced network configurations'.<br><br>The trace load can be reduced with this byte, e.g. if a deeper trace memory for GSM is wanted. | *255 (decimal), all modes active* |
| byte | 70 | Unnamed byte 70, SIP trace mode<br><br>1: Trace received frame if parse error<br>2: Trace unexpected received messages<br>4: Trace received messages unconditional<br>8: Trace transmitted frames<br>Code this byte as sum of the single modes. The trace load can be reduced with this byte, e.g. if a deeper trace memory for GSM is wanted. | *255 (decimal), all modes active* |
| byte | 75 | Unnamed byte 75, overrun LAN chip settings (Since firmware 3.00 via menu)<br><br>Hex 80: 10 Mbit/s, half duplex<br>Hex 81: Negotiation 10/100Mbit/s, half/full duplex<br>Hex 82: 100 Mbit/s, half duplex<br>Hex 83: Negotiation 10/100Mbit/s, half/full duplex<br>Hex 84: 10 Mbit/s, full duplex<br>Hex 85: Negotiation 10/100Mbit/s, half/full duplex<br>Hex 86: 100 Mbit/s, full duplex<br>Hex 87: Negotiation 10/100Mbit/s, half/full duplex<br>By default 10 Mbit/s is used because in worst case with 30 active calls via AS551 there is a maximum LAN load of about 3 Mbit/s and we detect less lost packets via LAN, if we use lower speed between a LAN-switch and an endpoint (in this case AS55x). This value should only be changed, if the LAN-switch has any problem with the default setting (e.g. it does not support 10 Mbit/s). | *0, 10 Mbit/s, full duplex* |
| byte | 77 | Unnamed byte 77, SIP REFER acknowledge mode<br><br>0: Acknowledge with OK<br>1: Acknowledge with NOTIFY | *0* |

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| byte | 78 | Unnamed byte 78, offered telephone event payload type<br><br>According to SIP RFC, allowed payload type for SIP telephone event are 96 to 127, the AS55X normally offers 123. But some PBXs need a special payload type, offered by the AS55X, it can be set here. This value has no affect to call setup directions from the PBX to GSM. The SIP telephone event is needed for DTMF via SIP (RFC2833/4733). | *0 ( same as 123)* |
| byte | 82 | Unnamed byte 82, SIP phone context processing<br><br>The SIP phone context is an additional information element of the SIP FROM or TO headers in order to include number prefixes.<br>With the value 1 of this byte, phone context is used, if the phone number is not in national or international order.<br>With the value 2, phone context will be used unconditional and a national or international prefix of the phone number will be removed. | *0, phone context processing off* |
| byte | 87 | Unnamed byte 87,<br><br>If an access from GSM is not permitted, the call will normally be rejected immediately. But in some cases, e.g. in context with call forwardings, it might be possible to delay this rejection. A delay can be entered in seconds. | *0, immediately* |
| byte | 90 | TCP message distance<br>In SIP mode some PBXs loose TCP messages, if they follow each other too fast. With this byte a minimum message distance can be defined in 10ms steps. | *0* |
| byte | 91 | minimum DTMF tone length from GSM<br>The minimum tone length of DTMF tones from GSM that will be detected in steps of 4 ms. Changing of this setting is useful only in case of faulty interpretation of noise. | *0, same as 12 (48 ms)* |
| byte | 103 | Telnet option raw TCP mode<br>If this byte is set to 1, no Telnet control codes are used. This is mostly called 'raw TCP mode'. | *0* |
| byte | 107 | read own voice number trials<br>This byte specifies the trials of reading the voice number of the own SIM. If there is no number stored on the SIM, the startup of the module gets faster if this value is set to 0. | *5* |
| byte | 109 | pause between FAX call-setup trials<br>If a FAX call-setup failed, after this time in seconds, the WLL is trying again. | *0, same as 60* |
| byte | 110 | pause between FAX call-setup trials after connection<br>If a FAX connection failed, after this time in seconds, the WLL is trying again. | *0, same as 10* |
| byte | 113 | maximum call trials to local FAX<br>The maximal number of call trials to the local FAX device, that is connected to the analogue interface of the WLL. | *0, same as 3* |
| byte | 114 | maximum call trials to remote FAX<br>The maximal number of call trials to the remote FAX device, that is connected via GSM to the WLL. | *0, same as 3* |
| byte | 115 | minimum DTMF tone length from wired side<br>The minimum tone length of DTMF tones from ISDN or SIP that will be detected in steps of 4 ms. Changing of this setting is useful only in case of faulty interpretation of noise. | *0, same as 12 (48ms)* |
| byte | 118 | remove leading ‘0’ zero from calling number<br>If the PBX adds a leading '0' to the calling number in case the origin of the call is in the wired network and this digit leads into trouble with callback or detection of PBX internal and external numbers. It can be removed in dependence on the type of number. If this byte is 0, the earlier reparation of the prefix error will be maintained.<br>If this byte is 1, a leading zero will be removed from an unknown number.<br>If this byte is 2, a leading zero will be removed from an ISDN national number.<br>If this byte is 4, a leading zero will be removed from an international number.<br>If this byte is 7, a leading zero will be removed from all number formats.<br>A value of 128 deactivates this function. | *0, previous behaviour* |

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| byte | 121 | for EU-3/PH8 to allocate network type<br>Normally the EU-3 module preferably registers to an UMTS network. If byte 121 is set to 1, it exclusively registers to a GSM Network and if the byte 121 is set to 2, if exclusively registers to a UMTS network. This setting is globally for all channels. | *0* |
| word | 15 | Unnamed word 15, SIP local RTP port base<br><br>By default, the AS55X uses RTP port numbers beginning from 30000. If there should be a conflict e.g. due to port forwarding in the router, this port number base can be changed. | *30000* |
| word | 16 | Unnamed Word 16 to 18, Ethernet MAC address overrun | *All 0, factory setting* |
| word | 19 | Unnamed word 19, SIP REGISTER repetition period<br><br>If the AS55X is in SIP client mode, after the time value set here, the registration process at its server will be repeated. Set this value in seconds. | *Default: 300 (5 minutes)* |
| word | 20 | Unnamed word 20, SIP OPTIONS repetition period<br><br>The AS55X periodically sends an OPTIONs message, the period length can be changed here. In SIP peer mode, the sending begins immediately after startup, in SIP client mode after successful registration at the server. In SIP server mode the AS55X sends this message to every registered client. The OPTIONs message can be used as keep alive function for NAT router entries. Set this value in 10 milliseconds, a period time of 0 will suppress the OPTIONs sending. The response from the endpoint can be ignored, even if it is negative. | *Default: 1000 (10 seconds)* |
| word | 21 | Unnamed word 21, service port<br><br>This is the listen port for service access via LAN. The default port is 29999 (TCP). | *Default: 0 (same as 29999)* |
| word | 22 | Unnamed word 22, jitter buffer initial delay<br><br>If the jitter in a network should be very high, the adaptation can result in some audio disturbance of VoIP connections. Before increasing this value in bad audio cases, it should be evaluated with a network protocol tester, if the jitter is really the cause. This value is understood in ms, the allowed range is 20 to 250. | *Default: 0 (same as 50 ms)* |
| word | 24 | Unnamed word 24, repetition period of NAT keep alive messages<br><br>This value is understand in 10 ms steps. | *0 (same as 1000, 10 seconds)* |
| word | 26 | Unnamed word 26, outgoing local TCP port range begin | *Begin:10000* |
| word | 27 | Unnamed word 27, outgoing local TCP port range end<br><br>With SIP TCP, the local port for outgoing TCP connections is changed because a port is blocked some time after terminating a connection. The local port range can be set with these words. If word 27 is smaller than word 26, always the local SIP port will be used. | *End: 10999* |
| long word | 10 | Unnamed long word 10, SIP CLIP source header list.<br>The CLIP from the PBX, used by AS55X for callback issues, can be present in different SIP headers. And the effective CLIP entry of these headers can differ. With this long word, the sequence for searching the CLIP can be defined. Each digit of this long word represents a SIP header for searching the CLIP.<br>Digit value 1: From header<br>Digit value 2: P-Asserted-Identity header<br>Digit value 3: P-Preferred-Identity header<br>Digit value 4: Contact header<br>Digit value 5: Remote-Party-ID<br>Other values are not defined.<br>The processing of this list begins from the right bound digit. By default the AS55X first searches the CLIP in the P-Preferred-Identity header. If this is unsuccessful (e.g. the header does not exist), the P-Asserted-Identity header will be checked. If also no success, the AS55X looks into the FROM header. | *00000123* |