

## **AS55X Release notes**

### **Firmware and Service Software 3.00**

The Firmware 3.00 is the junction of ISDN release 1.11 and SIP release 2.29. The switching between ISDN and SIP can be done with the service software. This firmware can be used for all AS55X hardware versions.

#### **1. New and improved features**

##### **1.1. LAN speed and simplex/duplex settings and watching**

Physical LAN settings can now be set via configuration platform and the actual used or negotiated operation mode can be watched with 'View LAN status'.

##### **1.2. Additionally add prefix '0' to CLIP to wired interface**

Some PBXes need this prefix in order to manage call back properly.

#### **2. Problem solution**

##### **2.1. Fixed link of ISDN B-channel and GSM channel failed**

This initially present function was lost with the firmware release 1.05 and is reactivated now.

##### **2.2. Suppress DHCP request as connection check**

If the LAN interface is a DHCP client, DHCP requests are additionally used as connection check to the network. This function can be suppressed by setting of unnamed bit 35.

### **Firmware and Service Software 3.01**

#### **1. New and improved features**

##### **1.1. Upgrade with upgrade code**

Upgrading for new features is now possible with an upgrade code. The installation window 'Enter upgrade code' generates an upgrade key. This key must be sent to the manufacturer for generation an upgrade code. The received upgrade code has to be copied into the dedicated field and then applied.

##### **1.2. Test licence**

Every AS55X gets a test licence for maximum device features for a duration of four weeks. With updating to firmware 3.01, the test licence is automatically granted. Please read further information in the help context of the configuration.

## **1.2. DialCommand**

The DialCommand server for working with SIP on the wired domain of the AS55X is implemented.

## **2. Problem solution**

### **2.1. Fix false number coding from SIP**

Sometimes the AS55X is getting called or calling party numbers like +0123 or +00123 which are obviously wrong coded. These numbers are now by default converted to 0123 respectively 00123. If this recoding should cause problems, it can be deactivated by resetting the unnamed bit 4.

### **2.1. Enhanced Via header evaluation**

In some cases with proxies, the SIP-response from the AS55X seems to come to nothing. The evaluating of the received Via header in SIP is enhanced now for proper working with proxies.

## **Firmware and Service Software 3.03**

### **1. New and improved features**

#### **1.1. Second peer IP address**

A secondary peer IP address can be defined now for a backup PBX. Call setups to the PBXs are sent to the primary peer IP address except if the connection to this address is not given but the connection to the second peer IP address is present. Call setups in direction to GSM are accepted from both IP addresses.

#### **1.2. GSM cell info available in multiplexer mode**

In earlier releases, cell information was not available if Telnet was activated. Now both can be used at the same time. If Telnet is used for internet access via GPRS, cell info can decrease data throughput a bit. Because of the high trace traffic that reduces the record history, cell info should only be requested, if necessary for troubleshooting.

#### **1.3. Sent short messages in statistics**

The number of successfully sent short messages via Telnet is shown in the statistics.

#### **1.4. GSM PBX implemented**

The GSM PBX provides facilities for SIP DID registration or multiplexed single user registrations at a SIP provider. SIP numbers can directly be linked to GSM numbers or DTMF dialling is possible with or without speed dialling lists in both directions.

## **1.5. Get calling party number from Remote-Party-ID header**

The default search sequence for determining the calling party number from SIP headers is now:

- P-Preferred-Identity
- P-Asserted-Identity
- Remote-Party-ID
- From
- Contact

## **1.4. SIP UPDATE implemented**

## **2. Problem solution**

### **2.1. Enhancements in SIP Record-Route processing**

There was a problem regarding interworking with SIP providers resulting in an abort of the connection after about 30 seconds. This problem is fixed.

### **2.2. Ignore Trying response from registrar**

It may happen that the registrar responses with a Trying provisional response to the REGISTER request. This will be ignored now.

### **2.3. PRACK configuration**

To increase flexibility, the PRACK sequence request can be configured now. It may be possible to request it always or never in order to activate early media or to avoid one way audio.

### **2.4. Password handling**

In earlier releases, passwords are included in the configuration data set. Because of privacy, they are handled extra now and there are only place holders included in the configuration data set that is saved into a file or uploaded from a target device. If the firmware of a device with passwords in its configuration is updated to this release or a higher one, the configuration must be uploaded and downloaded again in order to transfer the configuration into the new format.

### **2.5. IP address and port redirection**

Some PBXes cannot handle the SIP connection to an AS55X at the LAN properly if it is also registered at a SIP provider and set an IP address and port redirection to the public values. By setting of bit 67 this redirection can be ignored.

# Firmware and Service Software 3.04

## 1. New and improved features

### 1.1. SIP over TCP

TCP is now available as additional transport protocol for SIP. UDP remains the default transport protocol as SIP has its own mechanism to avoid data loss. The TCP protocol can be chosen with the extended SIP settings.

### 1.2. Speed dialling list for GME

If GME and GSM-PBX is used, the speed dialling list from GSM is available for GME dialling as well.

### 1.3. SIP INFO for supervision of connection

Some PBXes do not handle reINVITES for connection supervision properly. With this release, the INFO method is selectable alternatively for this issue.

### 1.4. VLAN tagging

VLAN tagging is implemented now. A VLAN tag with VLAN ID and priority can be defined in the extended SIP settings form. This tag is valid for SIP over TCP and UDP and for RTP.

### 1.5. Pass GSM CLIP in SIP client mode

Normally a SIP client can send only his own user number or username as CLIP to the provider. But in special cases the AS55X is wanted to include the GSM CLIP as calling party number into the INVITES to the provider. This facility is implemented now.

### 1.6. Receive dial able GSM numbers in SIP client mode

The AS55X in client mode can now receive a dial able GSM number. All received numbers that cannot be assigned to the GSM PBX link list and that are not the clients local username can be treated as direct dial able GSM number.

## 2. Problem solution

### 2.1. Device reset in ISDN mode

There was a problem regarding sending of multiple ISDN frames in very fast succession. This fault concerns the firmware releases 3.00 to 3.03, it is fixed.

### 2.2. Enhancements of supervision mechanism in PassThrough mode

There was a problem with trunk checks by the PBX in PassThrough mode. They were caused by ISDN STATUS messages outside of a connection, but with connection dependent call references. These messages are passed through now. In this context enhancements were made for supervisory to avoid blocking of B-channels.

# Firmware and Service Software 3.05

## 1. New and improved features

### 1.1. Recorder for Audio from GSM

Sometimes there were complaints about bad audio quality or unreliable DTMF reception over GSM. Audio from the GSM network can be disturbed, distorted or interrupted and there can be noise or echo. For better classification of such complaints, an audio recorder is implemented. The recorded signal will be saved as standard WAV file.

### 1.2. DTMF-relay

Additionally to DTMF converting between LAN and GSM via telephone-event in the RTP stream (RFC2833/4733), the processing via dtmf-relay over SIP signalling is implemented. Both of them can be activated independently.

### 1.3. 7e1 Emulation

For GSM circuit data purposes, sometimes the 7-e-1 character framing is used, but the inserted GSM engines do not support this framing. The 7e1 emulation may help in such cases. Please ask your distributor to recognise the necessity and how to configure the emulation.

### 1.4. Telnet Enhancements

Some virtual Com Port drivers really want to control a remote hardware via Telnet protocol. The AS55X responds to such control codes since this release. Additionally a raw TCP mode is implemented that does not expect and not use any Telnet control characters. The raw TCP mode can be activated by setting unnamed byte 103 to 1.

## 2. Problem solution

### 2.1. Message distance with TCP

In some cases, PBXes do not handle SIP messages over TCP correctly if they follow one another too fast. A minimum distance of SIP messages can be configured now with unnamed byte 90 in 10ms steps from 0 (value 0) to 150ms (value 15).

### 2.2. DHCP

If the AS55X works in DHCP client mode, sometimes there was a lack in complete registering the received IP address at the server. This problem is solved.

### 2.3. SIP User-Agent header

Since firmware version 3.03 the SIP User-Agent header is enhanced with device type and hardware release. Some PBXes do reject some characters of these values. These ones are replaced.

# **Firmware and Service Software 3.10**

## **1. New and improved features**

### **1.1. BRI interface control**

From this firmware release, the AS550 and AS551 ISDN BRI devices will be supported.

### **1.2. SNMP implemented**

Lots of device and statistics information can now be read from the AS55X by SNMP. Additionally the statistics of the AS550 can be restarted. The correspondent MIB file, needed by the SNMP manager, can be found in the subfolder 'Documents' of the service software.

### **1.3. Fix subscriber number in national or international order**

Some ISDN PBXes cannot handle incoming destination numbers with ISDN unknown type. The fix subscriber numbers can now begin with a 'N' or an 'I' character to deliver ISDN national or international number format to the PBX.

### **1.4. Accept tel uri**

In single cases, SIP PBXes use a numbering scheme with Tel URIs instead of SIP URIs. These ones are accepted now by the AS55X.

### **1.5. All files in one installation packet**

Since version 3.10, all files of a new release are included in the installation packet. The firmware can be found in the subfolder 'Firmware' and the release notes in the subfolder 'Documents' of the service software. The advantage is, all related files can now be found at one location, especially as the amount of files will increase.

### **1.6. Short message server**

A short message server is implemented for easy sending and receiving short messages via the AS55X through network applications. Such application named 'SMS Center' is offered. Interested parties, who want to write SMS applications their selves, are requested to contact the distributor for API information about the message server.

### **1.7. SMS callback request**

If a call setup to GSM remains unsuccessful, it is possible to send a predefined short message to the called participant - normally a callback request. This can take place automatically or on demand.

# **Firmware and Service Software 3.11**

## **1. New and improved features**

### **1.1. WLL control**

From this firmware, the control of the new WLL device is implemented.

### **1.2. SMS callback request enhancements**

Deviating from the global settings for sending a short message callback request, they now can be defined individually via the 'Individual call-setup' form.

The own voice number of the GSM channel is needed for SMS callback request purposes but it sometimes is not present at the designated location on the SIM. It will now be additionally read from the phonebook entry one if the name matches exactly this text: 'ownvoicenumber'. The detection is case insensitive.

### **1.3. SIP P-Asserted-Identity header**

Normally a SIP provider cannot handle different numbers in the From header. In order to make them able to get the CLIP from GSM, the P-Asserted-Identity header can now be included.

## **2. Problem solution**

### **2.1. SIP CSeq header**

In firmware version 3.10 there was an error in the CSeq header that can result into abort of call-setups from GSM to SIP. This error is fixed.

### **2.2. ISDN BRI PassThrough in PTP mode**

Some wired network providers do not support all ISDN layer 2 messages in point-to-point configuration. This will no longer cause a interface reset of the AS55X.

## **Firmware and Service Software 3.12**

### **1. New and improved features**

#### **1.1. Detection of PBX internal and external numbers**

A detection is implemented, if a number, exchanged with the PBX, is a number of the PBX domain (extension number) or not.

By default, the AS55X assumes a wired number to be an extension number of the PBX if the calling number is no national and no international number. If the PBX also sends calling numbers in national or international order, there can be made exceptions with the PBX domain number list. That means, the AS 55X will treat all there defined number-beginnings as extension numbers of the PBX.

#### **1.2. SMS callback request enhancements**

Different texts can be defined now if the origin of the call is within the PBX domain or not. The selection of the text to be sent depends on detection of PBX internal and external numbers.

The calling party name is now available for the SMS callback request for SIP calls.

#### **1.3. Add a leading '0' to the dialed number into the PBX**

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, if the bit 65 is set.

#### **1.4. Remove a leading '0' from the calling number from the PBX**

If the PBX adds a leading '0' to the calling number if 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 now. For this function, the byte 118 must be set to '7'.

#### **1.5. Display own SIM number if CLIP from wired side is present**

This new function leads the information through, if the caller wants a number to be displayed or not. The feature can be activated in globally 'General GSM settings' and individually in 'Individual call-setup list'.

#### **1.6. DTMF code for GSM channel restart**

For AS55X WLL there is a new DTMF code that restarts one specific or all GSM channels. This makes the GSM channel register preferably onto the home network of the SIM. This code is '#734#' for all channels, '#7341#' for the first and '#7342#' for the second channel.

#### **1.7. Dial command with ISDN**

The extended dial command feature is now available for ISDN BRI and PRI devices of the AS55X. If the AS55X is working in PassThrough mode and if a 'Code number for dialling from GSM to wired network' is specified, this one can precede the dialled number to ISDN in order to select the direction of the call.

#### **1.8. Save call setup only from PBX domain**

It is possible now, to avoid save call setups for external call sources from the PBXs point of view. That are mostly calls from the wired network to an extension that is forwarded to a GSM number (via the AS55X).

## **1.9. PassThrough backup**

If the connection to fixed network is lost, all calls can be routed via GSM. For this feature, a licence is necessary.

## **1.10. TC63i and EU-3 in AS551**

Now the wireless module types TC63i and EU-3 are possible in AS551 slave PCB. Usage of EU-3 will be necessary if only contact exits to an UMTS network.

Note: Telnet and SMS API is not available with EU-3 !

## **1.11. Splash SMS**

The SMS API is enhanced for sending of Splash SMS. Please refer to the latest version of SMS API description for the dedicated command.

## **1.12. Fore EU-3 to dedicated network type**

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.

## **2. Problem solution**

### **2.1. WLL: Immediate work after device reset**

After a device reset with the reset button it is no longer necessary to download a configuration with the service software. Standard phone and FAX operations are working immediately. But if the SIM requests a PIN, it must be entered by the DTMF command via the connected phone.

### **2.2. Duplication of read short messages via short message server**

There are conditions that can result into duplication of read short messages via the short message server in devices that use the GSM module TC63i. This problem is fixed.

### **2.3. Instability after reception of a flood of broadcast messages**

After reception of a great mass of broadcast messages (more than 200) without any gap, the AS55X could get instable. This had affect to all LAN functions, VoIP audio stream was chopped in that case. This problem is fixed with firmware 3.12.