AS55X Series

SERVICE MANUAL

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1   General functions

1.1   Fax via GSM

It is possible to send and receive faxes over the GSM mobile phone network using a custom G3-fax machine. You can use the WLL550 Pro gateway with integrated FXO or any AS55x/WLL550 VoIP gateway with an analog telephone adapter via a dedicated SIP-channel.

Normally fax transmissions over the GSM network are critical because the handshake is not optimized fir it. For better results the WLL550 Pro buffers the transmission process and splits it into two phases. First the fax document will be submitted from the sending machine to the WLL550 Pro's memory and then time-delayed from the WLL550 Pro memory to the receiving fax station. The WLL550 Pro will try up to three attempts to submit a fax. During this time the channel LED is flashing continuously and no further call is possible via this channel. However a fax transmission can fail for some reason and by splitting the transmission it might happen that the first phase of the transmission is successful but the second phase fails. To ensure that the sender gets information about the unsuccessful transmission the WLL550 Pro can generate a receipt for all successful and failed fax transmissions via a response fax.

For using the fax service a SIM card with activated GSM-fax service is mandatory. Ususally you will get a separate fax number from your service provider (also called multi-numbering). All incoming fax calls will be forwarded automatically to the analogue fax port or the SIP fax-channel. Outgoing faxes will be forwarded via a GSM-channel (UMTS and LTE do not support the GSM fax service).

For programming instructions see 5.5.

1.2   Callback

Due to the restrictions of displaying the CLIP within GSM networks ('CLIP -no screening' is not available and only a GSM number can be transmitted) a called party does not know which PBX subscriber is calling via the AS55X or who has initiate the call-setup. The called subscriber however, cannot reach the call initiator for a call back, because all subscribers of the PBX could have initiated this call. For this purpose the AS55X can record calls or call setup data.

All call-setups could be saved, so the AS55X can backtrack every call setup's origin. The called party initiates a call to the transmitted CLIP number (one GSM number of the AS55X) and by using of the call-setup list the callback can be routed the original subscriber of the PABX. Please consider that only the last call to one destination can be saved. A previous existing call-setup will always be overwritten by the new one. It is mandatory that both sides transmit their number (CLIP).

For programming instructions see 5.8.2, 5.8.3, 5.10.2 and 7.5.

1.3 Callback termination

The callback termination function is implemented to save fees on the mobile phone site. It is possible to call the AS55X with or without transmitting your caller ID (CLIP / CLIR). The call-setup or the call will be aborted immediately and the AS55X will initiate a callback to the mobile phone.
There a few different possibilities:

- 1st: the incoming call will be rejected by the AS55X
  (free of charge, callback via CLIP)
- 2nd: the incoming call has to be rejected manually by the mobile phone
  (free of charge, callback via CLIP)
- 3rd: the call will be rejected after entering a code
  (NOT free of charge, callback via DTMF code)

For programming instructions see 5.10.1, 5.8.2 and 5.10.3.

1.4 Short message callback request

To improve the callback options the 'short message callback request' function is implemented to supply a called subscriber with additional call information. If you call a customer via GSM you can choose between number presentation (CLIP) or number restriction (CLIR). In case of CLIR is active the customer sees a number of the AS55X but could not know who is the subscriber of the calling number. With 'short message callback request' it is possible to send a SMS short message after a call with additional call information. Two different text templates with up to 160 characters can be defined. The selection of the text to be sent depends on the calls origin is within the PBX domain or not. The text phrases are always the same but can be complemented by the following variables:

- Calling number (SIP or ISDN number of calling party)
- GSM channel number as displayed
- GSM channel number in national order
- Freely entered number via DTMF.

It is possible to send this short message automatically or controlled by the customer via DTMF code. The calling party name is available for the SMS callback request for SIP calls.

For programming instructions see 5.8.5

1.5 GSM channel groups

The routing function of the AS55x does not work via single GSM channels, because of the higher channel number. Single channels can be combined to a group, but a group might also consist of only one channel. Therefore, if the user does not want to work with groups, he can simply assign one channel to each group (the configuration for GSM groups can do such assignment automatically). Normally a GSM channel group consists of channels, registered to the same GSM network. Routing functions of the AS55x point to groups. The selection of a channel within a group is alternating, i.e. all channels of a group statistically will have the same load.

One GSM channel can be a member of different groups, e.g. one group points to a VPN and another one points to a GSM network.

For programming instructions see 5.9.3.

1.6 VPN

Virtual private networks (VPN) in GSM can significantly decrease costs, especially if GSM channels of an AS55x are members of VPNs. Further characteristics of VPNs are that members of a VPN can call each other via short user numbers without network prefixes, the CLIP from member to member is displayed as such short user number as well.

The AS55x provides a special support for a comfortable access to GSM channels, which are members of VPNs. In order to separate VPN user numbers from subscriber number at the PBX, a VPN prefix has to be created for each GSM channel of the AS55X that is part of the VPN. From the sight of the PBX, this GSM-VPN is an independent network with its own prefix. For a dialling to a user of this network, the VPN prefix has to lead the VPN user number. All routing functions of the AS55x can be used via this VPN prefix. For a call-setup from a VPN member to the PBX, the AS55x adds the VPN prefix, so the PBX subscriber can identify the caller as a VPN member. Callbacks will be routed correctly because of the consequent use of the VPN prefix.

Example: The GSM channel 3 of an AS55x is a member of a VPN with the user number 4444. The created VPN prefix for this GSM channel is +49123 (the VPN prefix has to be defined in international order, here with the German country code). A PBX subscriber wants to call the VPN member 5555, so he can dial 01235555 if the location of the PBX is in Germany as well. The appropriately configured AS55x will route this call via GSM channel 3. If this VPN member is calling a PBX subscriber himself, his number will also be displayed as 01235555.

If there is much traffic with a GSM VPN, multiple channels of an AS55x may get the same VPN prefix. These channels are combined to a GSM channel group and the routing function of the AS55x chooses one ready channel of this group if a call-setup to this VPN is made.

For programming instructions see 5.8.3, 5.10.2, 5.10.3, 5.11.1 and 5.11.5

1.7   Internet

To fulfil the requirements for a mobile office use the *WLL550 Pro* can work as a broadband-internet access which a basic function of the *WLL550 Pro*. The related radio channel with an LTE or UMTS/HSPA+/HSDPA module is the radio channel one. A SIM card with high speed data capabilities is mandatory in this case. Please also check the details of your mobile phone contract for a special data tariff to avoid unexpected high charges.

The internet speed depends on various parameters, such as the type of your *WLL550 Pro*, the capabilities of your mobile phone network, the local field strength quality and the operating grade of the shared medium. Providing you have the optimal conditions an internet speed of 7,2 MB/s is possible with UMTS and up to 40 MB/s via LTE.

The *WLL550 Pro* connects to the internet, if valid APN information is configured manually or when the *'Go online'* box is checked in the *'Internet'* settings or the DTMF code *#721#* is entered on the POTS interface.

For the use with a single computer you can connect the *WLL550 Pro* directly to the LAN interface of the *WLL550 Pro*. But you need an external NAT-router if the internet access should be available for further network devices.

For programming instructions see 5.10.4.
For LAN setup look here 5.2

1.8   PassThrough

PassThrough is available for all AS55X with BRI and PRI ISDN interface. Therefore the AS55X will be looped into the connection between the PBX and the primary multiplexer respectively the base terminal. No additional interface at your PBX is needed. All connection in both directions will be supervised by the AS55X. Depending on the AS55X configuration, the call will be redirected to the GSM network or signalized on both media. Calls which are accepted on mobile phone can be redirected in the PBX. Calls from mobile phone can be handled over the AS55X and so the CLIP of you PBX extension will be transmitted.

First operation mode is "MobileTwinning". Meaning that all called numbers in direction to PBX will be compared with a number list within the AS55X. If a listed number is detected, the AS55X will initiate a parallel call to the mobile phone. So the PBX extension and the mobile phone will ring simultaneously. Depending on the call accepting the other channel will be closed.

The second operation mode is "Interception". Here all calls in direction from PSTN will be checked and all configured destinations will be routed via the GSM network. No channel to the PSTN will be used.

For programming instructions see 5.8.3, 5.10.3 and 5.11.1.
more about the PassThrough backup mode here > 1.14

1.9   SimSwitch

With a SimSwitch AS55X it is possible to use up to five SIM cards for each GSM channel. E.g. SIM cards with a special amount of free minutes (SMS) can be used. The AS55x checks if the maximum number of minutes is reached and switches automatically to the next SIM card. This will also work with a free amount of SMS. It is also possible to change the SIM card permanently in a circle

For programming instructions see 0.

1.10 Telnet

The AS55X is primarily designed as a voice device, but offers Telnet as an additional service.

Within the Telnet feature you have two operation modes:
First the "SMS/GPRS"-mode which can be used independently from the voice data and secondly the "Full Access"-mode which must be used alternately to voice data. With Telnet (and a virtual COM port or additional software) you can sent SMS, use GPRS or have full access to one of the channel. There are up to two (AS550) or six (AS551) Telnet channels available.

For programming instructions see 0 and 5.11.3.

1.11 SMS Server

Within the AS55X we created an interface to send SMS short messages directly from an application via the AS55X. You need no additional Telnet channel license, no virtual COM port driver and you can use up to 30 GSM channels of the AS551 for sending and receiving SMS short messages. If you want to implement the SMS features into your applications, ask your local dealer for "API for sending and receiving short messages via AS55X". This is a small proprietary API only for sending and receiving SMS via AS55x.

For programming instructions see 5.11.4.

1.12 GSM-PBX

With GSM-PBX it is possible to achieve a direct link between a VoIP number and your mobile phone. Your AS55X VoIP could be used as complete mobile PBX. Therefore you only need a sufficient broadband connection. Your AS55X will register at your SIP provider so all calls will be transmitted automatically to your mobile phone. The AS55X has to be SIP client. It is also possible to use DTMF dialling.

In the other direction you can use the AS55X to present your VoIP number. Therefore a call to the AS55X, followed by DTMF dialling is necessary.

For often used numbers it is possible to save some shortcuts.

For programming instructions see 5.7.1, 5.7.3, 5.7.4, 5.10.2, 5.10.3 and 5.11.5.

1.13 SNMP

To supervise your AS55X you can use a Service Software or SNMP (Simple Network Management Protocol). It is possible to configure two SNMP Manager with different rights. The AS55X supports SNMP version 1 with GET, SET and TRAP messages. For dedicated MIB file ask your local dealer. The following functions are supported:

- TRAP:
    - System reboot

- SET:
    - Statistic reset

- GET:
    - System information (Device Type, Serial number, Hardware release, Firmware release, Device name, Temperature, Number of GSM channel, ...)
    - System statistic (Total system runtime, number of system reboots, min./max. temperature, interface layer 2 errors, ...)
    - Wireless channel engine information (Serial number, firmware release, hardware release,...)
    - Wireless channel errors (engine out of order, SIM missing, PIN of SIM unknown, ...)
    - Wireless channel mode (Channel not available, engine starting up, call to GSM network, registered, voice call from GSM network, ...)
    - Wireless channel network information (registered network, signal quality, ...)
    - Wireless channel statistics (min. / max. signal quality, call attempts to GSM, calls to GSM connected minutes to GSM, ...)
    - SIM Switch information (active SIM, connected minutes for each SIM)

For programming instructions see 0.

1.14 PassThrough backup mode

All calls shall be routed via GSM if the connection to fixed network is completely lost or if the network cannot route the calls due to congestion or resource problems.

If "PassThrough backup" is activated a call proceeding tone or announcement will be heard if a PassThrough backup call is in progress.

Please keep in mind, that the call-setup time is longer in this case as the AS55X BRI tries to forward the call to the fixed network first. It checks via timer if all dialled digits did arrive and finally it starts the call via GSM.

It is strongly recommended to check, if this audio signal can be heard by the calling user in the case of a backup call as he should have knowledge about this special (and normally more expensive) routing.

For programming instructions see 5.11.7

1.15 ProviderSimSwitch

✋	The feature is only available for devices with the SimSwitch hardware

With powering up the WLL 550 Pro the "ProviderSimSwitch" starts testing the GSM networks by two criteria. First is the speed of the internet connection (UMTS, GPRS or no access). Second is the signal quality of the GSM network. If a full working network is detected (100% signal quality and full UMTS service) any further testing will be stopped immediately. The search will be repeated until a working GSM network is found.

The time out for each SIM card logging into a GSM network is 45 seconds after the restart of the GSM module. For testing the APN setup there is an additional period of 30 seconds to connect to the internet.

Finally the "ProviderSimSwitch" will remain at the SIM with the best available network. If a network connection is lost for more than 10 seconds or if a new search is triggered manually, the network search will restart considering the previous network data.

For programming instructions see 5.11.8

1.16 DTMF Codes (WLL550 Pro)

In some cases instant access to the setup of the WLL 550 Pro is necessary. For this you can use a simple touchtone phone on the analogue TEL-port (POTS) of the WLL 550 Pro.

- to enter a PIN by DTMF, use the following code:

    #*1*xxxx# for SIM of channel 1 and
    #*2*xxxx# for SIM of channel 2.

    xxxx is the place holder for the PIN number.

- to force a GSM channel to register preferably onto the home network of the SIM. You can cause a restart of one specific or of all GSM channels by using the following DMTF codes:

    #734# (REG) reset of all GSM channels
    #7341# reset of GSM channel 1
    #7342# reset of GSM channel 2

- to separate outgoing fax calls from voice calls, you have to dial a fax prefix (e.g. *8 004944130005…) for each fax. This prefix (*8) is not a part of the called number will be deleted by the WLL 550 Pro.

- for triggering a network search manually (ProviderSimSwitch)

    #77#   for all GSM channels
    #77n#   for a specific GSM channel n = 1 or 2

    The DMTF input will be acknowledged by a special tone.

- The DTMF code of the WLL550 Pro for internet access has been enhanced to allow data roaming.

    #723#' for internet access with data roaming.

    The already known code '#721#' keeps its function, internet access without data roaming. Both codes are irrelevant, if internet access is configured with the service software. Please keep in mind, your SIM contract must permit data roaming as well!

- Enabling the start hash codes for WLL550 Pro (FW 3.16)

  It is possible now to enter start hash codes e.g. to switch call forwarding in GSM networks. All codes beginning with a star can be entered normally. But all codes beginning with a hash must be entered in a special manner as the beginning with a hash will result in a conflict with internal DTMF commands of the WLL.

  For codes beginning with one hash please enter *1#* . For codes beginning with a double hash please enter *2#* .

  E.g. to switch call forwardings off, *2#002#* has to be entered.

  To enable this feature, unnamed **byte 9** must be set to **0** and unnamed **bit 75** must be **set** (>>> *Special settings*).

- Simplified internet access of the WLL550

  As long as internet access is not set by service software, the access can be enabled by the DTMF code *#721#* and disabled by the code *#720#* .

- To select a SIM card of the *'ProviderSimSwitch'* manually

  #79nm#   *n* is the place holder for the channel (1 or 2)
                             m is the place holder for the SIM- card (1 or 2)

  Example: *#7921#* activates SIM card 1 of channel 2

2   Technical Features

2.1   AS551 Master (PRI/VoIP)

- Power supply:

  Built-in power supply
  Voltage: 90 – 264V AC, 47 – 440 Hz
  IEC power connector within two 2,5A fuses (medium time lag)
  Power input: 6 channel (MC55i):
                                 ca. 8W / ca. 17W (min/max)

- Radio modules:

  Gemalto/Cinterion MC55i
  Quad band GSM   850/900/1800/1900 MHz

  Gemalto/Cinterion TC63i/TC65i/TC65i-X
  Quad band GSM   850/900/1800/1900 MHz

  Gemalto/Cinterion EU3-E
  Dual band UMTS   900/2100 MHz
  Dual band GSM   900/1800 MHz

  Gemalto/Cinterion PH8
  Quad-band GSM   850/900/1800/1900MHz
  Five-band UMTS   800/850/AWS/1900/2100MHz

  Gemalto/Cinterion AGS2
  Quad band GSM   850/900/1800/1900 MHz

  Gemalto/Cinterion PHS8-P
  Quad-band GSM   850/900/1800/1900MHz
  Five-band UMTS   800/850/AWS/1900/2100MHz

  Gemalto/Cinterion PLS8-E
  Five Band LTE   800/900/1800/2100/2600 MHz,
                         FDD-Band (20,8,3,7,1)
  Tri Band UMTS (WCDMA)
                         900/1800/2100 MHz, FDD-Band (8,3,1)
  Dual Band GSM   900/1800 MHz

  Antenna: 1x antenna splitter 6 to 1 - SMA female connector
                         3V Mini-SIM cards

- Interfaces:

  LAN: Ethernet 10/100 Base-T, RJ45
  USB: USB Type B
  PRI: ISDN main, RJ45, S_{2M} Output
        ISDN aux, RJ45, clock synchronization, PassThrough input
  Slave Connector, RJ45

- Protocols:

  SIP2.0 UDP/TCP, client-, server-, peer-mode
  DHCP, DNS, STUN, ARP, VLAN, TELNET, SNMP
  DSS1, QSIG

- Codecs:

  G.711 a-law/µ-law

- Maintenance:

  Service software (https://www.kuhnt.com)
  Interfaces: local: USB (security: password)
                       remote: LAN, GSM (security: username and password)

- Size:

  19 inch rack aluminium (AS551) / 1 rack unit (1U)
  Size: width: 483mm, height: 45mm, depth: 300mm
  Weight: AS551 Master:
                    ca. 2,5kg

- Environment

  Operating temperature: 5°C to 40°C
  Storage temperature: -10°C to 70°C
  Humidity: 0% to 90%

2.2   AS551/4V BRI

- Power supply:

  Built-in power supply
  Voltage: 90 – 264V AC, 47 – 63 Hz
  IEC power connector within two 2,5A fuses (medium time lag)
  Power input: 4 channel (MC55i):
                                    ca. 5W / ca. 12W (min/max)

- Radio modules:

  Gemalto/Cinterion MC55i
  Quad band GSM   850/900/1800/1900 MHz

  Gemalto/Cinterion TC63i/TC65i/TC65i-X
  Quad band GSM   850/900/1800/1900 MHz

  Gemalto/Cinterion EU3-E
  Dual band UMTS   900/2100 MHz
  Dual band GSM   900/1800 MHz

  Gemalto/Cinterion PH8
  Quad-band GSM   850/900/1800/1900MHz
  Five-band UMTS   800/850/AWS/1900/2100MHz

  Gemalto/Cinterion AGS2
  Quad band GSM   850/900/1800/1900 MHz

  Gemalto/Cinterion PHS8-P
  Quad-band GSM   850/900/1800/1900MHz
  Five-band UMTS   800/850/AWS/1900/2100MHz

  Gemalto/Cinterion PLS8-E
  Five Band LTE   800/900/1800/2100/2600 MHz,
                                  FDD-Band (20,8,3,7,1)
  Tri Band UMTS (WCDMA)
                                  900/1800/2100 MHz, FDD-Band (8,3,1)
  Dual Band GSM   900/1800 MHz

  Antenna: 1x antenna splitter 6 to 1 - SMA female connector
                                  3V Mini-SIM cards

- Interfaces:

  LAN: Ethernet 10/100 Base-T, RJ45
  BRI: 2x ISDN TE (ISDN S0 Slave), RJ45, clock synchronization, PassThrough input, ISDN subscriber
        2x ISDN NT (ISDN S0 Master), RJ45, NT simulation, PassThrough output
  Reset button

- Protocols:

  DHCP, DNS, STUN, ARP, VLAN, TELNET, SNMP
  DSS1, QSIG

- Codecs:

  G.711 a-law/µ-law

- Maintenance:

  Service software (https://www.kuhnt.com)
  Interfaces: remote: LAN, GSM (security: username and password)

- Size:

  19 inch rack aluminium (AS551 / 4V BRI) / 1 rack unit (1U)
  Size: width: 483mm, height: 45mm, depth: 300mm
  Weight: AS551 Master / 4V BRI:
                         ca. 2,2kg

- Environment

  Operating temperature: 5°C to 40°C
  Storage temperature: -10°C to 70°C
  Humidity: 0% to 90%

2.3 AS551/2V BRI

- Power supply:

    Built-in power supply
    Voltage: 90 – 264V AC, 47 – 63 Hz
    IEC power connector within two 2,5A fuses (medium time lag)
    Power input: AS551 2 channel (TC63i):
                                    ca. 2,5W / ca. 6W (min/max)

- Radio modules:

    Gemalto/Cinterion MC55i
    Quad band GSM   850/900/1800/1900 MHz

    Gemalto/Cinterion TC63i/TC65i/TC65i-X
    Quad band GSM   850/900/1800/1900 MHz

    Gemalto/Cinterion EU3-E
    Dual band UMTS   900/2100 MHz
    Dual band GSM   900/1800 MHz

    Gemalto/Cinterion PH8
    Quad-band GSM   850/900/1800/1900MHz
    Five-band UMTS   800/850/AWS/1900/2100MHz

    Gemalto/Cinterion AGS2
    Quad band GSM   850/900/1800/1900 MHz

    Gemalto/Cinterion PHS8-P
    Quad-band GSM   850/900/1800/1900MHz
    Five-band UMTS   800/850/AWS/1900/2100MHz

    Gemalto/Cinterion PLS8-E
    Five Band LTE   800/900/1800/2100/2600 MHz,
                        FDD-Band (20,8,3,7,1)
    Tri Band UMTS (WCDMA)
                        900/1800/2100 MHz, FDD-Band (8,3,1)
    Dual Band GSM   900/1800 MHz

    Antenna: 1x antenna splitter 6 to 1 - SMA female connector
                        3V Mini-SIM cards

- Interfaces:

  LAN: Ethernet 10/100 Base-T, RJ45
  BRI: ISDN TE (ISDN S0 Slave), RJ45, clock synchronization, PassThrough input, ISDN subscriber
  ISDN NT (ISDN S0 Master), RJ45, NT simulation, PassThrough output
  Reset button

- Protocols:

  DHCP, DNS, STUN, ARP, VLAN, TELNET, SNMP
  DSS1, QSIG

- Codecs:

  G.711 a-law/µ-law

- Maintenance:

  Service software (https://www.kuhnt.com)
  Interfaces: remote: LAN, GSM (security: username and password)

- Size:

  19 inch rack aluminium / 1 rack unit (1U)
  Size: width: 483mm, height: 45mm, depth: 300mm
  Weight: AS551 Master / 4V BRI:
  ca. 2,1kg

- Environment

  Operating temperature: 5°C to 40°C
  Storage temperature: -10°C to 70°C
  Humidity: 0% to 90%

2.4 AS551 Slave

- Power supply:

    Built-in power supply
    Voltage: 90 – 264V AC, 47 – 440 Hz
    IEC power connector within two 2,5A fuses (medium time lag)
    Power input: 8 channel (MC55i):
                            ca. 7W / ca. 18W (min/max)

- Radio modules:

    Gemalto/Cinterion MC55i
    Quad band GSM   850/900/1800/1900 MHz

    Gemalto/Cinterion TC63i/TC65i/TC65i-X
    Quad band GSM   850/900/1800/1900 MHz

    Gemalto/Cinterion EU3-E
    Dual band UMTS   900/2100 MHz
    Dual band GSM   900/1800 MHz

    Gemalto/Cinterion PH8
    Quad-band GSM   850/900/1800/1900MHz
    Five-band UMTS   800/850/AWS/1900/2100MHz

    Gemalto/Cinterion AGS2
    Quad band GSM   850/900/1800/1900 MHz

    Gemalto/Cinterion PHS8-P
    Quad-band GSM   850/900/1800/1900MHz
    Five-band UMTS   800/850/AWS/1900/2100MHz

    Gemalto/Cinterion PLS8-E
    Five Band LTE   800/900/1800/2100/2600 MHz,
                        FDD-Band (20,8,3,7,1)
    Tri Band UMTS (WCDMA)
                        900/1800/2100 MHz, FDD-Band (8,3,1)
    Dual Band GSM   900/1800 MHz

    Antenna: 1x antenna splitter 6 to 1 - SMA female connector
                        3V Mini-SIM cards

- Interfaces:

  Master connector, RJ45

- Size:

  19 inch rack aluminium (AS551 Slave) / 1 rack unit (1U)
  Size: width: 483mm, height: 45mm, depth: 300mm
  Weight: AS551 Slave:
                                 ca. 2,4 kg

- Environment

  Operating temperature: 5°C to 40°C
  Storage temperature: -10°C to 70°C
  Humidity: 0% to 90%

2.5   AS550 BRI/VoIP

- Power supply:

    External power supply
    Voltage: 7,5V – 30V DC / max. 1A
    Coaxial power connector ⊖―€―⊕
    Power input: approx. 3W / 6W (min/max) Minwa MM065P300CGS

- Radio modules:

    Gemalto/Cinterion BGS2-E
    (dual-band GSM) for 900/1800MHz

    Gemalto/Cinterion BGS2-W
    (quad-band GSM) for 850/900/1800/1900MHz

    3V Mini-SIM cards

    Antenna: 1x antenna splitter 2 to 1 - SMA female connector

- Interfaces:

    LAN: Ethernet 10/100 Base-T, RJ45
    BRI: ISDN TE (ISDN S0 Slave), RJ45, clock synchronization, PassThrough input, ISDN
            subscriber
            ISDN NT (ISDN S0 Master), RJ45, NT simulation, PassThrough output
    Reset button

- Protocols:

    SIP 2.0 / UDP as server, client or peer-to-peer
    Additional protocols: ARP, DHCP, DNS, STUN, VLAN, SNMP, TELNET
    Authentification: MD5

- Codecs:

    G.711 a-law/µ-law

- Maintenance:

    Service software (https://www.kuhnt.com)
    Interfaces: remote: LAN, GSM (security: username and password)

- Size:

  Desktop aluminium box

                      width: 153mm, height: 35mm, depth: 185mm
  Weight: ca. 750g without power supply

- Environment

  Operating temperature: 5°C to 40°C
  Storage temperature: -10°C to 70°C
  Humidity: 0% to 90%

2.6   WLL 550 Pro (wireless local loop)

- Power supply:

  External power supply:
  Voltage WLL550 Pro: 12V to 30V DC
  >> Micro Mate-N-Lok connecting cord included <<
  Power consumption: 2 channel (TC63i / EU3-E):
  Min. approx. 4,5W / max. ca. 10W
  (With Mascot type 9619)

- Radio modules:

  Gemalto/Cinterion TC63i/TC65i/TC65i-X
  Quad band GSM/CSD/GPRS/EDGE   850/900/1800/1900 MHz

  Gemalto/Cinterion BGS2-W
  Quad band GSM/CSD/GPRS/EDGE   850/900/1800/1900 MHz

  Gemalto/Cinterion PLS8:
  Five Band LTE   800/900/1800/2100/2600 MHz,
                                                 FDD-Band (20,8,3,7,1)
  Tri Band UMTS (WCDMA)   900/1800/2100 MHz,
                                                 FDD-Band (8,3,1)
  Dual Band GSM/GPRS/EDGE   900/1800 MHz

- Antenna: One antenna combiner 2 in 1 - SMA female connector
                       One auxiliary SMA connector for LTE diversity

- 3V mini-SIM cards

- Interfaces: LAN/Ethernet 10/100 Base-T, RJ45, 'auto sensing'
                                Two analogue FXS (POTS): FXO for fax and phone (CLIP),
                                 RJ11

- Network protocols: DHCP, DNS, ARP

- Maintenance: Service software *AS55X-Service*
                                for WINDOWS operating systems
                                (Download link: https://www.kuhnt.com/down/55xwin32.zip)
                                Interfaces: LAN, Reverse IP or GSM CS-data
                                (Secured by: username and password)

- Size/weight: Aluminium desktop case.
                               Approx. width 125 (154*) x height 35 x depth 185 mm
                               * with mounting bracket
                               Approx. 810 g (2 radio modules)

- Environment: Operating temperature: 5°C to 40°C
                              Storage temperature: -10°C to 70°C
                              Humidity: 0% to 90%

3   Exterior view / Connectors

3.1   AS 551 Master (PRI/VoIP)

- Front
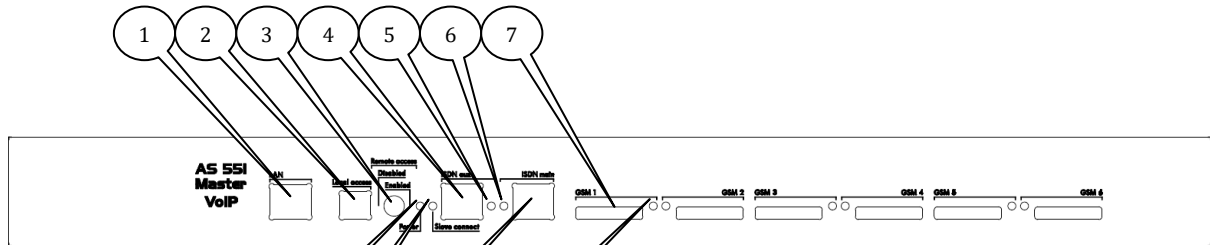
- Back
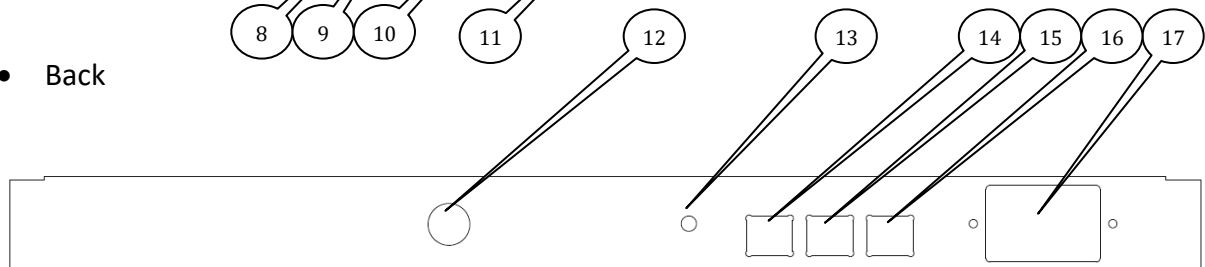
1. LAN connection
   10/100 Mbps connector for LAN (for SIP communication and configuration)
2. USB connection
   USB Type B for local access
3. Remote access (enable/disable)
   Switch to allow or deny remote access (LAN/GSM)
4. ISDN aux*
   RJ 45 connector for clock synchronisation or PassThrough input
5. ISDN aux LED*
6. ISDN main LED*
7. SIM Slot
   Depending on version; with two, four or six slots for 3V Mini-SIM cards
8. Power LED
9. Slave LED*
10. ISDN main*
    RJ 45 connector for S_{2M} output
11. Channel status LED
12. Antenna connection
    SMA connector (female)
13. Ground connection
14. Slave 3 Connector (channel 23-30)**
15. Slave 2 Connector (channel 15-22)
16. Slave 1 Connector (channel 7-14)Fehler! Textmarke nicht definiert.
17. IEC power connector for build in power supply

> RJ45 connectors for up to three AS551 Slaves (use a patch cable with a length of max. 0,5m)

---

* not for AS 551 Master VoiP
** not available for SimSwitch AS 551 Master (VoiP) / 2V and SimSwitch AS 551 Master (VoiP) / 4V

3.2 AS 551 / 4V BRI

- Front
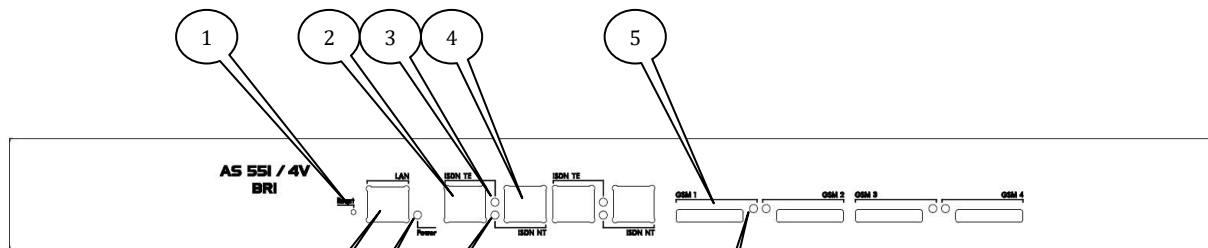
- Back
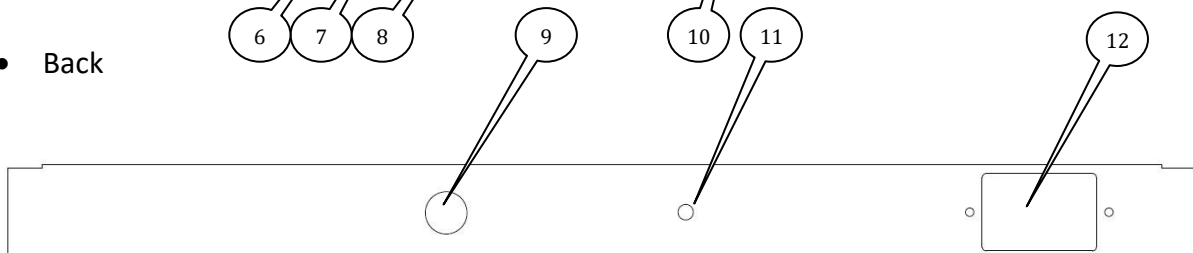
1. Reset button
   To restore the configuration press the reset button first and then plug in the power supply.
   After you release the button, the device will start with default configuration.
2. ISDN TE (ISDN Slave)
   RJ 45 connector for clock synchronisation, PassThrough input or ISDN Subscriber
3. ISDN TE LED
4. ISDN NT (ISDN Master)
   RJ 45 connector for PassThrough output or NT-Simulation
5. SIM slot
   Four slots for 3V Mini-SIM cards (first SIM of the channel)
6. LAN connection
   10/100 Mbps connector for LAN (for SIP communication and configuration)
7. Power LED
8. ISDN NT LED
9. Antenna connection
   SMA connector (female)
10. Channel status LED
11. Ground connection
12. IEC power connector for build in power supply

3.3 AS 551 / 2V BRI

- Front
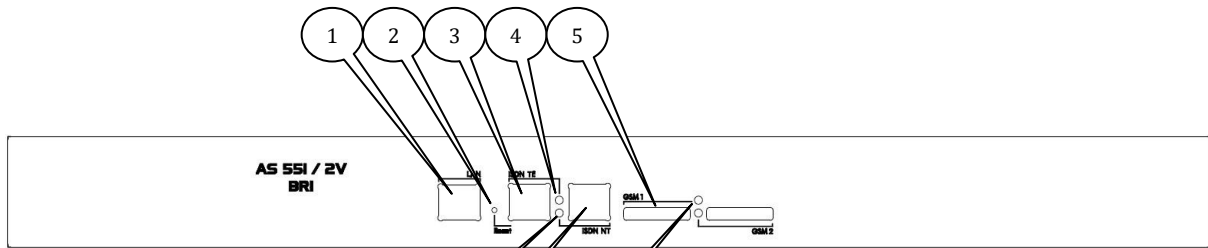
- Back
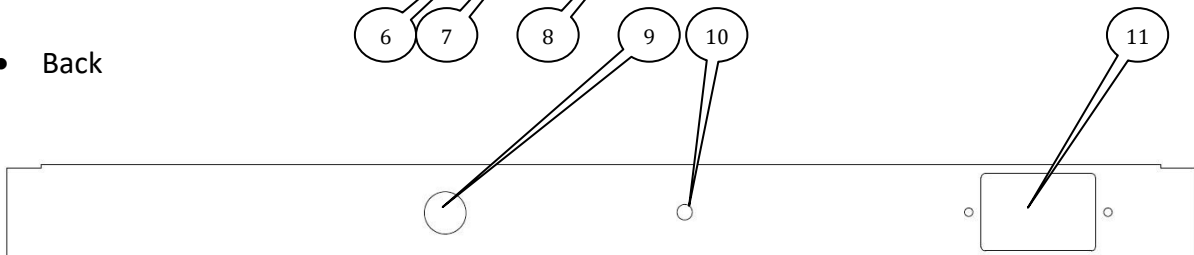
1. LAN connection
   10/100 Mbps connector for LAN (for SIP communication and configuration)
2. Reset button
   To restore the configuration press the reset button first and then plug in the power supply.
   After you release the button, the device will start with default configuration.
3. ISDN TE (ISDN Slave)
   RJ 45 connector for clock synchronisation, PassThrough input or ISDN Subscriber
4. ISDN TE LED
5. SIM slot
   Two slots for 3V Mini-SIM cards (first SIM of the channel)
6. ISDN NT LED
7. ISDN NT (ISDN Master)
   RJ 45 connector for PassThrough output or NT-Simulation
8. Channel status LED
9. Antenna connection
   SMA connector (female)
10. Ground connection
11. IEC power connector for build in power supply

3.4 AS 551 Slave

- Front
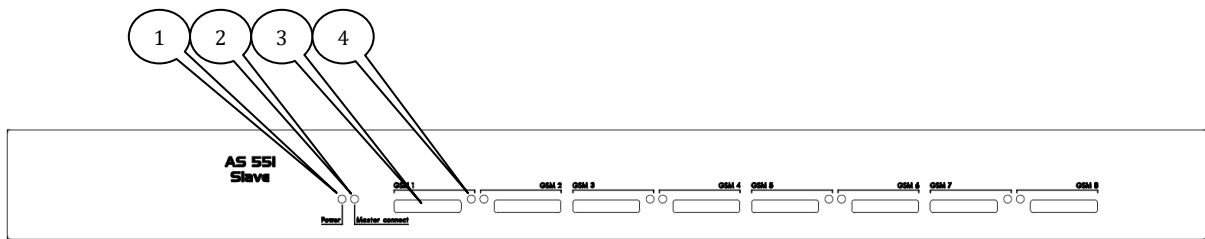
- Back
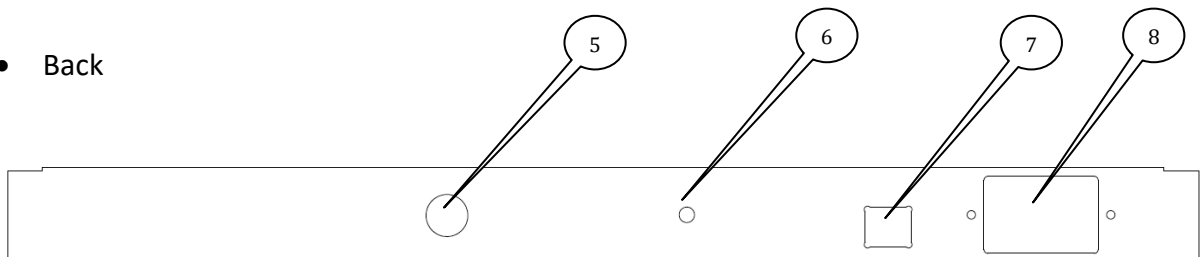
1. Power LED
2. Master LED
3. SIM Slot
   Expansion with up to eight slots for 3V Mini-SIM cards.
4. Channel status LED
5. Antenna connection
   SMA connector (female)
6. Ground connection
7. Master
   RJ 45 Connectors for the AS551 Master (max. 0,5m)
8. IEC power connector for build in power supply

3.5 AS550 BRI/VoIP

- Front
- Back
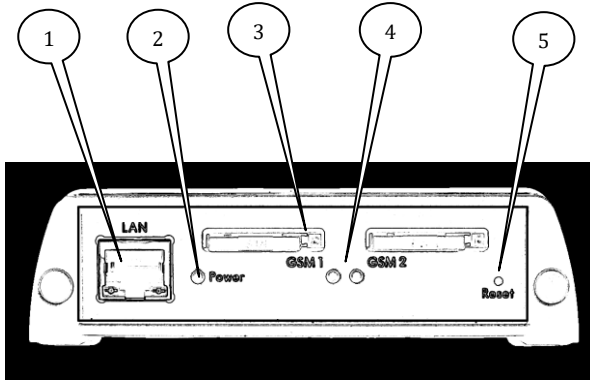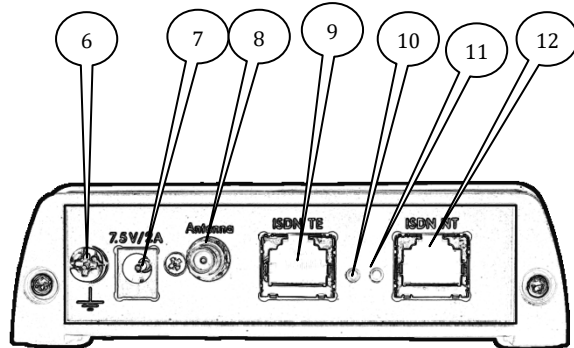
1. LAN connection
   10/100 Mbps connector for LAN (for SIP communication, SMS and configuration)
2. Power LED
3. SIM Slot for *GSM 1* and *GSM 2*
   Two slots for 3V Mini-SIM Cards.,
4. Channel status LED
5. Reset button
   To restore the configuration press the reset button first and then plug in the power supply.
   After you release the button, the device will start with default configuration.
6. Ground connection
7. Power connector for external power supply
8. Antenna connection
   SMA connector (female)
9. ISDN TE (ISDN Slave)
   RJ 45 connector for clock synchronisation, PassThrough input or ISDN Subscriber
10. ISDN TE LED
11. ISDN NT LED
12. ISDN NT (ISDN Master)
    RJ 45 connector for PassThrough output or NT-Simulation

3.6 WLL550 Pro (wireless local loop)
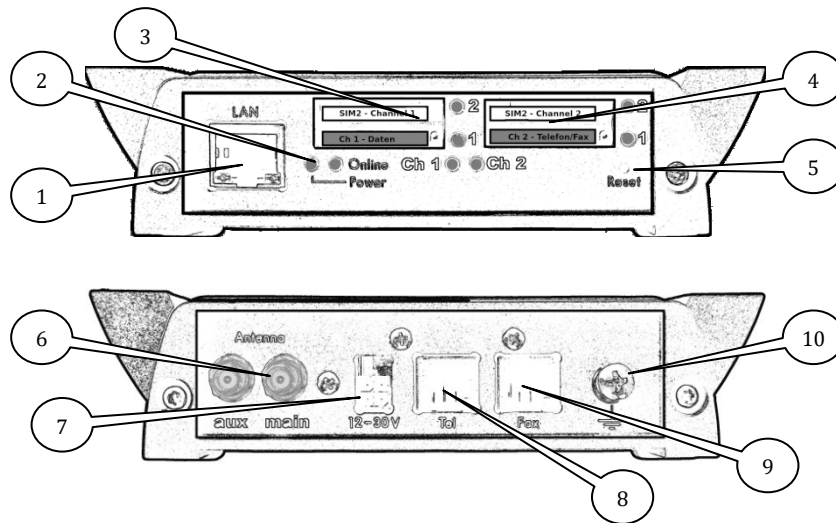
1. Network connector: LAN 10/100 Mbit/s
   (For configuration setup and internet access)
2. Power LED / Online status LED
3. Dual SIM card reader with status LED – channel 1   (LTE/UMTS/i-net, phone)
4. Dual SIM card reader with status LED – channel 2   (GSM/fax, phone)
5. RESET button
6. SMA connector main antenna, auxiliary SMA connector for LTE diversity
7. Power supply: 12V – 30V DC
8. Telephone-/PBX-connector (FXO)- RJ11
9. G3 fax connector (FXO) - RJ11
10. Earth terminal, GND

3.7 LED status

- Power LED

  o LED is steady green if the power supply is connected and the first configuration is done

  o LED is blinking red if the power supply is connected and no configuration is found (factory default settings by reset button)

  o LED is off if no power supply is connected

- Slave LED

  o LED is steady green if one slave device is detected and one of its channels is active

  o LED is steady red if one or more slave channels are configured but no slave device is connected.

  o LED is off if no slave channels are configured

- Master LED

  o LED is steady green if the master is detected

  o LED is steady red if no master is detected

- ISDN aux LED

  o LED is steady green if PassThrough or synchronisation is configured and a PRI input or sync signal is found

  o LED is steady red if PassThrough or synchronisation is configured but no PRI input or sync signal is found

- ISDN main LED

  o LED is steady green if there is a ISDN connection

  o LED is steady red if there is no connection or the connection is incorrect (e.g. a fault polarity)

- ISDN TE LED

  o LED is steady green if PassThrough or synchronisation is configured and a BRI input or sync signal is found

  o LED is red if there is no connection or an error is occurred

  o LED is blinking green if ISDN TEI exchange was successful but ISDN layer 2 is down (only in ISDN point-to-multipoint mode)

- o LED is blinking yellow if ISDN layer 2 is active but there has been no TEI exchange before (only in ISDN point-to-multipoint mode) (e.g. conflictive point-type configuration)

- ISDN NT LED

    - o LED is steady green if PassThrough or synchronisation is configured and a BRI input or sync signal is found

    - o LED is red if there is no connection or an error is occurred

    - o LED is blinking green if ISDN TEI exchange was successful but ISDN layer 2 is down (only in ISDN point-to-multipoint mode)

    - o LED is blinking yellow if ISDN layer 2 is active but there has been no TEI exchange before (only in ISDN point-to-multipoint mode) (e.g. conflictive point-type configuration)

- SimSwitch LED

    - o LED is steady green if the dedicated channel is selected by the SimSwitch

    - o LED is off if the channel is configured but currently not in use

    - o LED is blinking green if the channel is not configured

- Channel status LED

    - o LED is steady green if the channel is ready

    - o LED is blinking green if a call is active

    - o LED is steady red if the channel is switched off

    - o LED is blinking red one time if no module is installed or the installed module is damaged

    - o LED is blinking red two times if no SIM is insert or the insert SIM is not detected

    - o LED is blinking red three times if a SIM error occurred or the PUK is requested

    - o LED is blinking red four times if no PIN is stored but requested by the SIM

    - o LED is blinking red five times if the stored PIN is wrong

    - o LED is blinking red six times if a wrong PIN is enters twice

    - o LED is blinking red seven times if no registration is possible for more than two minutes. The module will restart then.

    - o LED is blinking red eight times if the designated network is permitted and no login is possible

- o LED is blinking red ten times if there is an error in the data exchange with the GSM module or due an unknown hardware problem

- o LED is blinking red eleven times if there is an error while loading the echo cancelling application respectively while checking "echo cancelling level"

- o LED is blinking red twelve times if the module will restart by GSM channel restart timer

- o LED is blinking red thirteen times if an error occurred within the multiplex mode ("Telnet mode")

- Online LED (WLL550 Pro only)

  - o LED is steady green if the internet connection (WAN) is successfully established

3.8 RESET button

With the RESET button you can delete all configuration and security data and restore an AS55X or a WLL550 Pro to the factory default settings.

To initialise to factory default settings, first disconnect the power supply. Then reconnect the power supply while the reset button is already pressed. Keep on holding it until the green power LED is on again. After releasing of the button the gateway will instantly restart with the factory default settings restored.

Hint! Using the RESET button will not only delete all configuration data, but also all security information, like PIN and passwords etc.!

-

With the firmware Rev 3.20 some service modes were introduced to the RESET button

If the gateway is powered on you can initialize restart it or a remote service session by pressing the RESET button.

If the RESET button is hold the power LED will start the following flashing sequence:

*> 2sec. off **> 2 sec. alternating red/green** > 2 sec. off > **2 sec. yellow** > 2 sec. off >*
***2 sec. red** > 2 sec. off > then turning to green again.*

Releasing the RESET button after a determined time causes a certain action like a ReverseIP remote service session, a warm restart or a cold restart.

- **flashing red/green** > starts a **ReverseIP-connection**** to the manufacturer's service unit via internet
- **flashing yellow** > causes a warm restart
- **flashing red** > causes a cold restart (all counters, the ARP-table etc. will be deleted)

Please note!
All AS551 VoIP and PRI gateways do not have any RESET button but instead the "Remote access " - switch on the front panel can initialize a "ReverseIP Session" too, by just switching it on and off again.

** ReverseIP-connections are remote service session to a manufacturer's service computer. To protect your data it is recommended to init a remote service session by appointment only. An internet access via mobile phone network or LAN is mandatory!

4 Service Software

For the configuration of an AS55X you usually need the AS55X-Service-Software. The software is available at https://www.kuhnt.com. Please download the latest version (55XWIN32.zip) from the software section. You can use this software for all types of AS55X and on most of the MICROSOFT operating systems: Windows 9x, NT, 2000, XP, VISTA, Windows 7, 8 and 10.

After the download of the software you have to expand the zip-file. The installation will be executed with the "setup.exe". For a hassle free installation it is recommended that all running applications will be closed, because the install setup cannot update files if they are in use.

On Windows NT, 2000, XP, VISTA, Windows 7, 8 and 10 operating systems you need administrator privileges to install the software and to expand the AS55X USB driver on your computer. (USB Drivers for WIN7/8 64bit must be installed manually).

The default installation directory is: C:\programs\AS55X_Service or C:\program files(x86)\AS55X_Service. To approve the directory, you can start the setup with the installation button. After this you can also check the program group before the installation will finally copy all files into the chosen directory.

For the USB driver installation see 4.1.1.1.

The latest version of the service software is necessary to use all features of the latest firmware. The recent firm- and software releases you will also find on our website under: https://www.kuhnt.com. The current version of your service software can be checked under the "About this program" button in the menu bar or it is also listed in the topic of the window. To check the firmware of your device, use the "View device data" menu (see 7.14).

An overview of the AS55X-Service-Software and a description of all functions you will find in the next step.
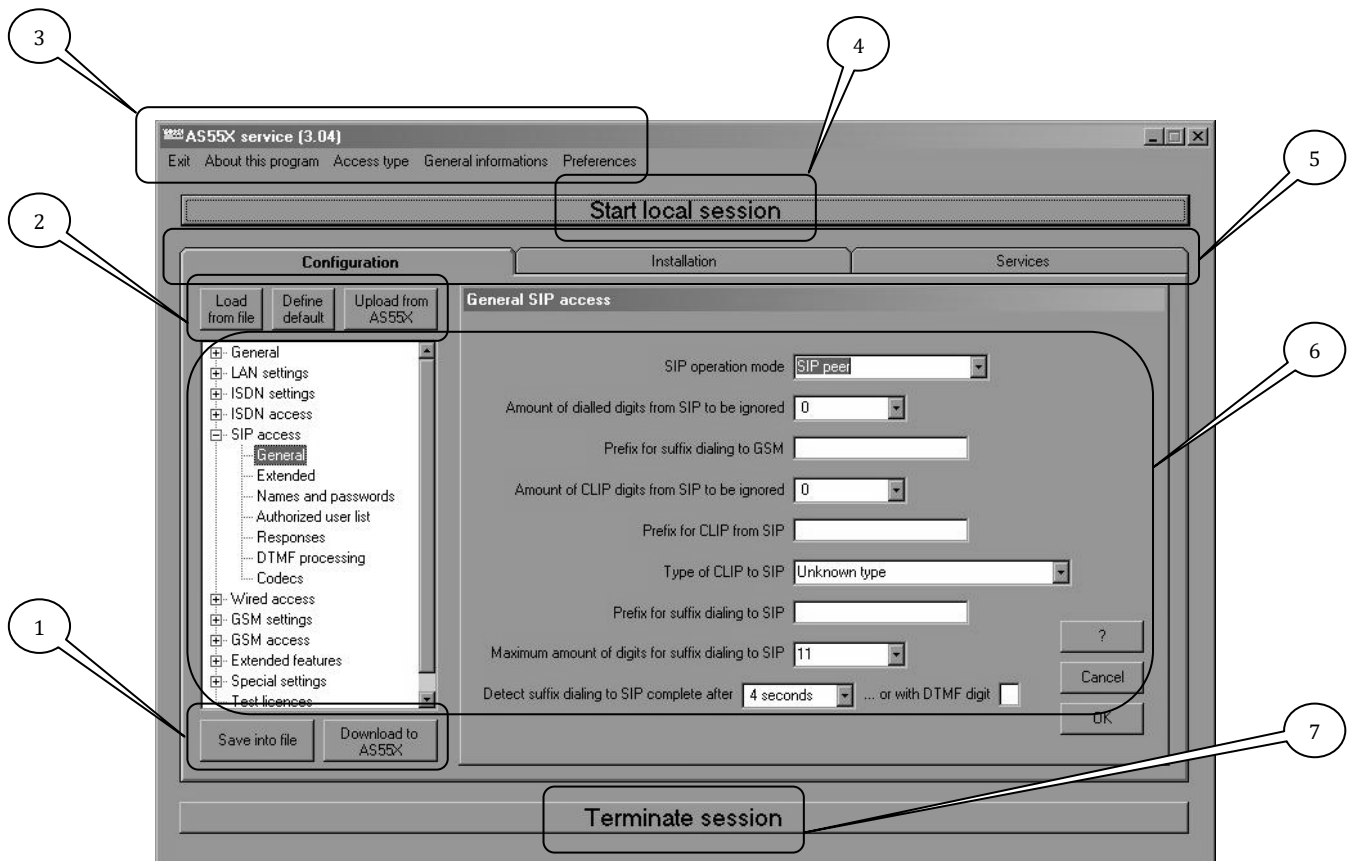
1. Buttons to save current configuration locally or to download configuration to AS55X
2. Buttons to load configuration from file, upload configuration from AS55X or to define a default configuration
3. Menu bar
   o Exit:
     Button to quit the program
   o About this program:
     To check date and number of software revision
   o Access type:
     To switch between the modes: local access (USB), access via LAN, access via GSM data and demo access
   o General information:
     Explanation about numbers, VPN and Telnet
   o Preferences:
     To activate Expert mode
4. Button to start the programming session (local, remote or demo)
5. To switch between configuration (see 4.1), installation data (see 6), and services (see 7)
6. Configuration or data area
7. Button to abort the current programming session

4.1 Access type

To connect to your AS55X for configuration and maintenance you have the possibility via USB, via LAN, via GSM-data or via Reverse IP/TCP. To edit setup data or for analyse trace data offline please use the "demo mode".

4.1.1 USB

Some AS55X are configurable via USB. But note that the USB driver is only able to detect one device at a time. It is not possible to configure more than one device at the same time. The local access is not protected by the remote access switch and only secured by a user/password request.

If you start a local session without installed USB driver, you will get a "runtime error". So please check USB driver installation first.

4.1.1.1 USB driver installation

The USB driver will be installed to the needed directory after the first start of the AS55X-Service-Software. Then the operation system will automatically find the driver.

Please note that in a Windows 7/8/10 32bit environment it is mandatory to start the AS55X-Service-Software in administrator mode for the first time. Therefore use "right click" on the menu item and choose "run as administrator". If you start the AS55X-Service-Software in user mode with left click, the USB driver will not be added to the system drivers. (USB Drivers for WIN7/8/10 64bit are available from the manufactorer and must be installed manually).

> ⚠️ Notice: You have to install the AS55X service software first <u>and</u> then start the application additionally once <u>before</u> you plug-in the USB jack of the AS5XX. Otherwise the device will not be detected!

4.1.2 GSM

If your network provides GSM CS-Data service it is possible to maintain a AS55X via remote fully featured. Configuration and forward traces are possible via data call. The access type GSM is classified as remote access and secured by username and password. All AS55X with a USB connection are additionally equipped with a physical remote access switch to control remote access.

> ⚠️ Notice: It is not possible to change the local access password via remote access.

4.1.3   LAN

LAN access is also classified as remote access and is secured with username and password. All AS55X with an USB connector are additionally equipped with a remote access switch to prohibit any unauthorized remote access. If you have a configured network interface with set IP address it is possible to configure the AS55X completely without changing your local IP settings.

After the AS55X is powered on you can use the search function of the configuration software to locate the AS55X in your network. All AS55X within the same collision domain will be listed in the configuration software with configured name, serial number and device type. This search works via broadcast. But be sure that your Windows system sends this broadcast via the correct LAN interface. Since the operating system Windows VISTA broadcasts are not send via all network interfaces. So if there is no search response it is possible that you use an active WLAN connection and the wired interface to connect to the AS55X simultaneously. In this case switch off all other network interface connections and try again.

If you select the listed AS55X, the AS55X-Service-Software tries to connect to the AS55X. If your network interface is configured and it is possible to reach the configured IP address (e.g. 192.168.0.2 by default, see textbox "Remote IP address") the connection will be established immediately. Otherwise you can configure a temporary IP address. This means that the configuration software detects that it is not possible to connect to the AS55X and you will be requested to set a temporary IP address for this session. Now you can set the IP address of your AS55X so that it is possible to connect to it. But note that this IP address will stay permanently and may concern SIP functions.

If all attempts are not successfully, connect via USB or use the reset button to restore the factory default settings (IP address 192.168.0.2).

⚠️	Notice: It is not possible to change the local access password via remote access.

4.1.4   Demo

If you want to edit a saved configuration, configure an AS55X setup offline or for analysing trace data you can use the demo mode. You can configure all types of the AS55X series. Start a demo session and select the AS55X type from the list.

4.1.5   Reverse IP access

For an advanced service it is possible to arrange a maintenance session with a remote computer by using a TCP connection. This session must be initialised by the AS55x or WLL550 gateway. The available physical connections are GPRS via GSM, UMTS, LTE or LAN, depending on the type of the gateway.

✋	By default this feature is not shown. If you want to use the Reverse IP access or uses the Reset-Button. Please contact your local dealer for further information.

4.2   Import / Export function

With the import and export function you can extract your list to a text file or you can copy the list to one of the following tables in the configuration program:

- "(Beginning of) GSM number" in menu "Wired access" / "Individual call-setup"
- "(Beginning of) GSM number" in menu "GSM access" / "By GSM number"
- "GSM number" in menu "Extended features" / "Mobile Twinning" / "Mobile Twinning list"
- "GSM number" in menu "GSM PBX" / "DTMF speed dialling list from SIP"
- "SIP number" in menu "GSM PBX" / "DTMF speed dialling list from SIP"

If you press "Export" the next window opens and you have to choose "yes" or "no". If you press "no", the whole list will be exported to a text file, which you can save anywhere. This file can only be imported into an AS55X the same type of table. If you press "yes", the listed GSM/SIP numbers will be copied to the clipboard. Now you can use these numbers in another menu.

If you press "Import" the next window opens and you have also to choose "yes" or "no". If you press "no", you can import a text file, which was exported from an AS55X. If you press "yes", your clipboard data from one of the listed menus will be inserted to your table.

5   Configuration
5.1   General

5.1.1   Tones and announcements

You can configure different dial tones and announcements for each direction.

Dialling tone ISDN/SIP -> GSM *default: continuous tone*	Dialling tone for call-setup direction from ISDN/SIP This tone or announcement is audible to the ISDN/SIP subscriber until the first digit of the suffix dialling is entered. In some cases it depends on the PBX whether this tone is switched through to the subscriber or not.
Dialling tone ISDN/SIP -> GSM if no GSM channel ready *default: pulsed dialling tone*	Dialling tone if no GSM channel ready This tone or announcement is audible to the ISDN/SIP subscriber until the first digit of the suffix dialling is entered and if no GSM channel is registered to a GSM network with sufficient signal quality. Only emergency calls can be made now. In some cases it depends on the PBX whether this tone is switched through to the subscriber or not.
Proceeding ISDN/SIP -> GSM *default: ringing tone*	Audible signal for confirming the selection of outgoing calls The AS55X is able to send a confirmation tone or an announcement to the ISDN/SIP subscriber to indicate that the dialling is completed. Useful, because the duration from ending the dialling process up to the moment the subscriber gets connected is much longer in mobile phone networks than in wired networks. In some cases it depends on the settings within the PBX whether this tone is audible to the subscriber or not.
Dialling tone GSM -> ISDN/SIP *default: special dialling tone 1*	Dialling tone for call-setup direction from GSM network This tone or announcement requests the calling subscriber from the GSM network to type in DTMF digits for suffix dialling.

Idle tone GSM -> ISDN/SIP *default: ringing tone*	Idle tone for call-setup direction from GSM network This tone or announcement is audible to the calling subscriber from the GSM network to indicate that the called phone is ringing after a DTMF suffix dialling or after a delayed fix subscriber call. Sometimes the GSM network adds some howling noise to the normal ringing tone. In this case the interrupted ringing tone is recommended.
Callback code entering tone *default: intermittent tone I*	Callback code entering tone This tone or announcement is audible to the GSM subscriber, when he is requested to enter the callback code.
Callback code acknowledge tone *default: intermittent tone II*	Callback code acknowledge tone This tone or announcement is audible to the GSM subscriber, when he has succeeded entering the callback code.
Minute reminder ISDN/SIP -> GSM *default: unchecked*	Minute reminder for call direction from ISDN to GSM If this box is checked and if the call-setup direction is from ISDN to GSM, every minute a short tone will be sent to the ISDN subscriber.

The following different variation of tones and announcements could be chosen. The announcements are available in Austrian, German, English, English 2 (Dutch accent), French, Dutch (Netherlands) and Finnish language. For personal announcements you could convert up to 25 messages with 260 seconds total length in common. For using personal announcements, ask you local dealer for the converting tool.

• Eurovision melody I
• Eurovision melody II
• "GSM gateway, please dial mobile phone number"
• "GSM gateway, please dial extension number"
• "Calling mobile phone, please hold the line"
• "GSM gateway, please enter callback code"

	sec	1	2	3	4	5	6	7

• Continous tone — 425 khz

• Special dial tone 1 — 425 khz

• Special dial tone 2 — 440 khz / 380 khz

• Pulsed dialing tone — 425 khz

• Intermittent tone I — 425 khz

• Intermittent tone II — 1336 khz

• Ringing tone — 425 khz

• Interrupted ringing tone — 425 khz

• Single short tone — 425 khz

• Busy tone — 425 khz

• Four busy tone — 425 khz

• Tone sequence — 1850 khz / 1450 khz / 950 khz
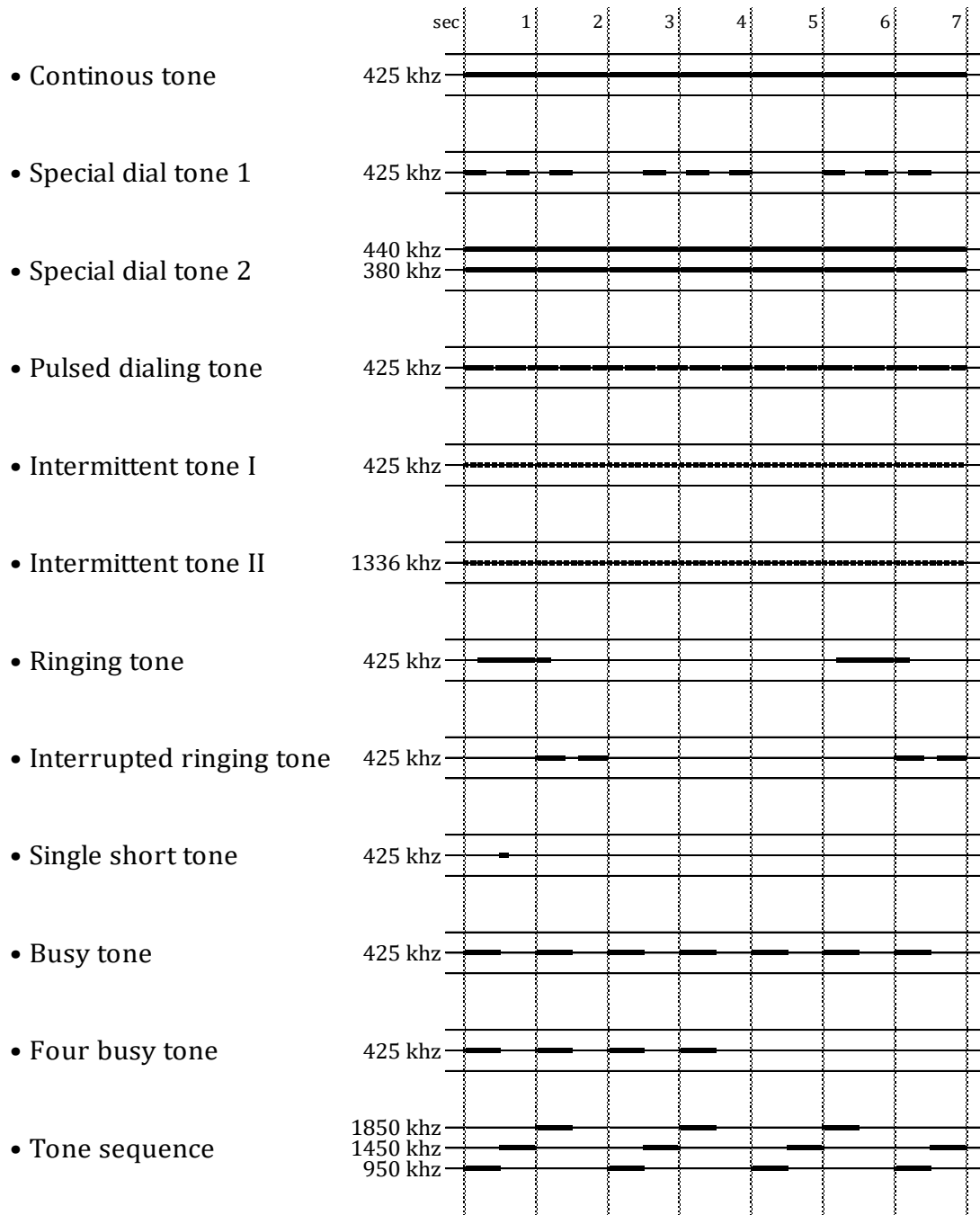
5.1.2   Volume levels

✋	This feature is not available in all SMS/data only gateways.

It is possible to change the volume level for audio and tones separately in each direction. You can adjust the level in 3dB steps. By default all values are set on 0dB level.

- Audio level ISDN/SIP -> GSM: Audio level directed ISDN to GSM
- Audio level GSM -> ISDN/SIP: Audio level directed GSM to ISDN
- Tone level -> ISDN/SIP: Tones and announcements directed to ISDN/SIP
- Tone level -> GSM: Tones and announcements directed to GSM

5.2 LAN settings

5.2.1 LAN interface Mode

Set the LAN interface operation mode in accordance to your requirements. There are four operation modes. The default value for all AS55x gateways is "Client at network without DHCP" with the IP address 192.168.0.2.

- Client at network with DHCP

  AS55X is a client on a DHCP based network. Network settings are found automatically supplied by an external DHCP-Server. Keep in mind that the IP configuration can change. Therefore this mode not suitable for SIP services, except the gateway is working as SIP client at the same time.

- Client at network without DHCP

  AS55X is a client on a managed network with fixed IP addresses. You have to know the IP addresses for the AS55X, the default gateway and the DNS server as well as the subnet mask in your network. This mode should be used, if the AS55X works in SIP Server or SIP peer mode.

⚠	Notice: If you configure a default gateway and DNS, take care that they are reachable. Otherwise the network component of the AS55X checks it and restarts the LAN interface if no gateway is available. This will cause that the LAN interface does not work. If you use a network without gateway and DNS, you do not need a subnet mask as well.

- Act as default gateway with DHCP and DNS

  AS55X is a DHCP and DNS server on a local, stand alone network. You can connect your phones by a switch and you have to configure them with automatic IP configuration. The IP address of the AS55X can be chosen however the subnet mask is automatically set to 255.255.255.0. The DHCP range and DNS Server name of the AS55X you can configure in "advanced network configuration".

⚠	Notice: Take note that all phones are configured in DHCP mode. The AS55X is not checking if a manually configured IP address is in use.

- Direct connection to endpoint

  AS55X is directly connected to the PBX without any network equipment between. Keep in mind that a "crossover cable" might be necessary. You only have to configure local IP address.

5.2.2   Advanced network configuration

- Remote STUN server name

    At this point you can configure an external STUN server. The AS55X must be set into client mode with or without DHCP. Also the IP address and default gateway must be entered in "LAN interface menu", otherwise the menu is greyed-out.

    If you use a network with NAT routing and your AS55X contacts your SIP provider, your outside global IP address is unknown. Normally your SIP provider is submitting this address in the SIP header. Otherwise the configured STUN server transfers your worldwide unique IP address to your gateway.

    For usage of the AS55X on local network a STUN server is not necessary.

    Default value: *not reported*

- Local DHCP server range begin / local DHCP server range end

    If your AS55X acts as "default gateway with DHCP and DNS", the range of the assigned IP addresses to the client can be set here.

    Default value: *The DHCP range starts one step behind the IP address of the AS55X and ends at the ends of the last IP address of the fourth octet. (X.X.X.254)*

- Local DNS server supported name

    If the AS55X acts as "default gateway with DHCP and DNS", it will serve DNS requests from clients to this name.
    Use the form: text1.text2, e.g. "as55xserver.as55xdomain" .

    Default value: *not reported*

- LAN physical settings

    Here you can switch between 10Mbps and 100Mbps in full or half duplex mode. On one hand 10 Mbps is enough bandwidth because in worst case with maximum number of active calls via AS55X there is a maximum LAN load of about 3 Mbps and there are less lost packets via LAN, if lower speed between a LAN-switch and an endpoint (in this case AS55X) is used. On the other hand there a problem may occur if you use 10Mbps because normally the other network clients use 100Mbps or 1000Mbps. Then the switch has to buffer the received messages from the other clients. If there is much traffic, this can cause an overflow and packets will be lost.

    Default value: *Auto negotiation*

- Trace received frames from LAN / Trace sent frames from LAN

  You can activate these options to save all network traffic. By default only relevant LAN messages will be traced. You can distinguish between sent and received traffic data.

  Though this setting will cause a lot of unnecessary trace data (especially audio data). It is not recommended for normal operation.

  Default value: *both unchecked*

- Trace received frames from LAN / Trace sent frames from LAN

5.3   ISDN PRI settings

⚠️ Notice: If you use the AS55X in ISDN mode, the SIP interface will be deactivated. The LAN interface can be used for programming only.

✋ ISDN PRI settings are only available for AS55X PRI devices.

5.3.1   Main interface settings

- Hierarchy:

  The PRI interface must be directly linked to the PBX and one unit has to be clock master and the other one has to be clock slave. The master has to generate a stable clock frequency and the slave must synchronize to it.

  Normally the PBX, the AS55X is connected to, gets its clock from the wired network. It is recommended to switch the PBX to master mode and the AS55X to slave mode. In this configuration, all units are working synchronously.

  If the PBX does not allow working as master at this interface and the AS55X must be clock master. The PBX will get two clocks, one from the wired network and one from the AS55X. Some PBXs cannot work with different clocks due to the offset between these clocks. This might cause problems with data and fax connections over the wired network or aborted connections.

  Therefore you can use the ISDN AUX Interface. This can be connected to the wired network interface of the PBX. To configure this option use the menu "synchron interface".

  Default value: *Slave*

  ✋ This option is greyed-out if the AS55X is working in PassThrough mode.

- Swap signal lines

  Normally the pins 1 and 2 of the RJ45 connector are the transmit lines of the master unit and the pins 4 and 5 are the transmit lines of the slave entity.

  By switching between master and slave mode, some PBXs swap these lines automatically, for some other PBXs jumpers must be set and others have fixed settings and this swapping has to be done within the wired installation.

  If this checkbox remains unchecked, the AS55X transmits on pin 1 and 2 in master mode and transmits on pin 4 and 5 in slave mode. If this checkbox is checked, the AS55X transmits on pin 4 and 5 in master mode and transmits on pin 1 and 2 in slave mode.

  Default value: *unchecked*

5.3.2   Main interface protocol

- Protocol

  You have to select the protocol of PRI-Interface. Two protocol types can be selected: DSS1 (defined by ETSI) or QSIG (defined by ECMA).

  Notice that both protocols are similar. It might work even with the wrong selection.

  Default value: *DSS1*

- Protocol option supervise connection:

  If an ISDN call is in active mode (connected), the AS55X can actively supervise the connection via ISDN STATUS requests every minute. This is recommended in all cases. If your provider or your PBX does not support this, the function must be disabled or the connection will be aborted exactly after one minute.

  Default value: *... by STATUS request (default)*

- Protocol option setup mode:

  After ISDN call-setup message of your PBX, the AS55X answer with different options. You can choose how the AS55X does it.

  o   CALL PROCEEDING; ALERTING with audio *(default value)*

  This is the default value and the version that is closest to the ETSI ISDN standard. The AS55X answer with ALERTING and generates the tones as listed in menu tones and announcements.

⚠️	Notice: In this case you can hear no tones and announcements out of the GSM network.

⚠️	Notice: Some PBXs generate the tones itself before ISDN CONNECT message. Therefore you cannot hear the AS55X generated tones and announcements.

  o   CALL PROCEEDING; ALERTING with ringing tone via GSM

  The AS55X answers with ALERTING after detecting ringing tones in GSM network. Not AS55X self generated tones can be heard, but tones and announcements of the GSM network are switched to the wired network.

⚠️	Notice: Some GSM networks use non standard tones. Therefore the tone detection can fail.

⚠️	Notice: Some PBXs generate the tones itself before ISDN CONNECT message. Therefore you cannot hear the AS55X generated tones and announcements.

o   CALL PROCEEDING; after Connection GSM: CONNECT

The AS55X does not answer with ALERTING after detecting ringing tones in GSM network. In this case the AS55X answers directly with CONNECT. This option shall suppress the ringing tone generation by the PBX.

⚠️	Notice: Some GSM networks use non standard tones. Therefore the tone detection can fail.

o   Immediately CONNECT

Some PBXs generate the tones itself before ISDN CONNECT message. With this setting, the AS55X answers immediately with CONNECT and all tones and announcements out of the GSM network are switched directly to the wired network.

⚠️	Notice: Be careful if you use your gateway in public network which is not free of charge. This choice will produce costs during the call-setup phase

- Protocol option reject

    If a SETUP from ISDN to the AS55X must be rejected (e.g. no GSM channel available), this can be done in two ways.

    The normal way is to reject the SETUP with RELEASE COMPLETE. But some PBXs misinterpret this as error.

    Alternatively the SETUP can be accepted and immediately disconnected.

    In most cases the system will work fine with both settings. If the PBX is wanted to reroute rejected calls, the success may depend on this protocol option.

    Default value: Reject by RELEASE COMPLETE

- Tone generation during call-setup

    During the call-setup phase the AS55X normally generates tones or switches audio from the GSM network to the ISDN interface. If the PBX should generate tones, it can be set here.

    This function is controlled by the progress indicator "in-band information available", but not all PBXs do interpret this indicator correctly.

    Default value: ... by AS55X

- Tone generation during call-termination

    During the call-termination phase the AS55X normally generates a busy tone to the ISDN interface. If the PBX is wanted to generate this tone, it can be set here.

    This function is controlled by the progress indicator "in-band information available", but not all PBXs do interpret this indicator correctly.

    Default value: *... by AS55X*

- CRC 4

    The PRI interface normally uses CRC 4 for synchronisation and error detection. If it does not use CRC 4 you have to switch off this option.

    Default value: *On*

5.3.3   Synchronisation interface (ISDN AUX)

- Synchronisation mode

    This option is greyed-out if the AS55X is working in PassThrough or slave mode. Otherwise you can choose the operation mode of the synchronisation interface. This is necessary if the PBX needs synchronised clocks at different primary rate interfaces.

    o   Off *(default value)*

    The AS55X generates its own clock.

    o   Additional terminal

    To synchronize the AS55X main interfaces clock to an existing PRI clock, connect the ISDN AUX interface parallel to your PBX (as an additional terminal) to the wired network (e.g. with an Y-adaptor). In this case the receive path of the connection are in high-impedance-state. The AS55X gets its clock rate from the wired network

⚠	Notice: If the ISDN AUX interface is not connected or the received clock rate is incorrect, the AS55X generates its own clock.

    o   Single terminal

    To synchronize the AS55X main interfaces clock to an existing PRI clock, connect the ISDN AUX interface to a free PRI interface of your PBX. The AS55X has to be exclusive participant (a single terminal) on this ISDN Bus. The AS55X gets its clock rate from the PBX.

⚠	Notice: If the ISDN AUX interface is not connected or the received clock rate is incorrect, the AS55X generates its own clock.

5.4 ISDN BRI settings

> ⚠️ Notice: If you use the AS55X in ISDN mode, the SIP interface will be deactivated. The LAN interface can be used only for programming.

> ✋ ISDN BRI settings are only available for AS55X BRI (ISDN2) devices.

5.4.1 Interface settings

- Hierarchy

    Every ISDN connection has exactly one master. This master simulates the clock. One or more slaves sync to the master signal. If the AS55X shall generate this clock, choose "Network, NT simulation" and plug your PBX into ISDN NT jack. Otherwise the AS55X is ISDN slave and you have to choose ISDN TE jack.

    Default value: Terminal, TE interface

    > ✋ This option is greyed-out if the AS55X is working in PassThrough mode.

- Point-type (line interface type)

    After you have chosen if the AS55X is ISDN master or ISDN slave, you have to choose if the BRI interface has to operate with a point-to-point or point-to-multipoint connection. From AS55X s point of view, a TE point-to-point interface (tie-line) is preferred. This one is the most flexible one and has no problems with synchronisation. If you choose point-to-multipoint TE connection, there is no possibility to dial a GSM number directly. In this case you have to use the MSN-GSM link list within a fixed combination between ISDN and GSM number. For point-to-point or point-to-multipoint NT connections you have to pay attention to the synchronisation.

    Default value: Point-to-point

- Synchronisation (ISDN AUX)

    In the ISDN network (NT) operation modes, the clock of the ISDN interface has to be generated within the AS55X. The AS55X produces a very stable and accurate clock, but as ISDN is a synchronous network, some PBXs need synchronous clocks at all interfaces to networks. Synchronicity faults can lead to connection aborts, disturbing noise or FAX and data transmission problems for example.

    The correct setting of the synchron interface (ISDN AUX)depends on the ISDN-line, the signal is picked off from. Preferably the synchron signal should be taken from a wired network interface of the PBX. If the PBX has only a PRI network interface,

synchron signal can be taken from an internal ISDN bus. In this case the PBX must unconditionally synchronise itself to the PRI interface.

First you have to check if the interface the AS55X synchronisation interface (ISDN TE) is connected to, uses a point-to-point or point-to-multipoint protocol. If the AS55X is the only connected terminal, please select "single terminal". Otherwise if more than the AS55X uses this interface use "additional terminal". For normal use one synchron interface is enough. All AS55X ISDN interfaces will synchronize to it. In special environments you can use the second ISDN TE interface to have a backup synchron interface. Then both interfaces have the same permission. The AS55X uses the interface where a clock is detected first. If this clock is lost, the other interface will be used.

A synchronisation will not be necessary if the PBX has no other ISDN network interface (e.g. VoiP).

Default Value: *Off*

🤚	This option is greyed-out if the AS55X is working in PassThrough mode.
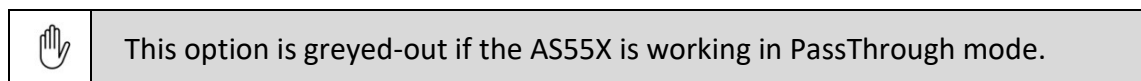
- Force layers up

    The clocking of an ISDN interface can be shut down. But in order to recognise interface failures and to react as fast as possible, keeping clock is recommended.

    If layer 1 is chosen, the AS55X will only force the interface clocking. In case of layer 1 and 2, address (TEI) management in point-to-multipoint modes is done as soon as possible and the data link layer will be held active.

    There is no additional power consumption of the AS55X if the lower layers are active.

    Default value: *Layer 1 and 2*

5.4.2   Protocol options

- Protocol

    You have to select the protocol of the BRI-Interfaces. Two protocol types can be selected: DSS1 (defined by ETSI) or QSIG (defined by ECMA).

    QSIG is originally defined for ISDN PRI interface. Here QSIG is used on BRI interface. But some PBXs use PRI elements (e.g. the channel-ID). For normal use set "DSS1" for point-to-multipoint connections and "DSS1 and QSIG" for point-to-point connections. In this case the AS55X will detect the used protocol after the first call setup from PBX to AS55X. If there are only connections from GSM to ISDN or the first call was from AS55X to PBX, a fixed setting of the concerned information elements is necessary. If the type, used by the PBX is unknown, it can be found in the trace of a call setup from the PBX.

    Default value: *DSS1*

- Protocol option supervise connection:

  If an ISDN call is in active mode (connected), the AS55X can actively supervise the connection via ISDN STATUS requests every minute. This is recommended in all cases. If your provider or your PBX does not support this, the function must be disabled or the connection will be aborted exactly after one minute.

  Default value: *... by STATUS request (default)*

- Protocol option setup mode:

  After ISDN call-setup message of your PBX, the AS55X answer with different options. You can choose how the AS55X does it.

  o CALL PROCEEDING; ALERTING with audio *(default value)*

  This is the default value and the version that is closest to the ETSI ISDN standard. The AS55X answer with ALERTING and generates the tones as listed in menu tones and announcements.

⚠	Notice: In this case you can hear no tones and announcements out of the GSM network.

⚠	Notice: Some PBXs generate the tones itself before ISDN CONNECT message. Therefore you cannot hear the AS55X generated tones and announcements.

  o CALL PROCEEDING; ALERTING with ringing tone via GSM

  The AS55X answers with ALERTING after detecting ringing tones in GSM network. Not AS55X self generated tones can be heard, but tones and announcements of the GSM network are switched to the wired network.

⚠	Notice: Some GSM networks use non standard tones. Therefore the tone detection can fail.

⚠	Notice: Some PBXs generate the tones itself before ISDN CONNECT message. Therefore you cannot hear the AS55X generated tones and announcements.

  o CALL PROCEEDING; after Connection GSM: CONNECT

  The AS55X does not answer with ALERTING after detecting ringing tones in GSM network. In this case the AS55X answers directly with CONNECT. This option shall suppress the ringing tone generation by the PBX.

⚠	Notice: Some GSM networks use non standard tones. Therefore the tone detection can fail.

  o Immediately CONNECT

Some PBXs generate the tones itself before ISDN CONNECT message. With this setting, the AS55X answers immediately with CONNECT and all tones and announcements out of the GSM network are switched directly to the wired network.

> ⚠️ Notice: Be careful if you use your gateway in public network which is not free of charge. This choice will produce costs during the call-setup phase

- Protocol option reject

  If a SETUP from ISDN to the AS55X must be rejected (e.g. no GSM channel available), this can be done in two ways.

  The normal way is to reject the SETUP with RELEASE COMPLETE. But some PBXs misinterpret this as error.

  Alternatively the SETUP can be accepted and immediately disconnected.

  In most cases the system will work fine with both settings. If the PBX is wanted to reroute rejected calls, the success may depend on this protocol option.

  Default value: Reject by RELEASE COMPLETE

- Tone generation during call-setup

  During the call-setup phase the AS55X normally generates tones or switches audio from the GSM network to the ISDN interface. If the PBX should generate tones, it can be set here.

  This function is controlled by the progress indicator "in-band information available", but not all PBXs do interpret this indicator correctly.

  Default value: ... by AS55X

- Tone generation during call-termination

  During the call-termination phase the AS55X normally generates a busy tone to the ISDN interface. If the PBX is wanted to generate this tone, it can be set here.

  This function is controlled by the progress indicator "in-band information available", but not all PBXs do interpret this indicator correctly.

  Default value: ... by AS55X

5.5   POTS (FXO) settings

✋	POTS settings are only available for WLL550 Pro.

5.5.1   Interface settings

- Operation mode / Fax call prefix

    There are two ways to use a fax machine with the WLL550 Pro, First you can attach your phone and fax directly to its dedicated connectors. The second way is to attach the WLL550 Pro to an analogue PBX or a Phone/Fax combo. In default mode all incoming faxes and calls will be directed to different POTS ports. In 'trunk mode' all calls are directed to the phone connector. To separate outgoing fax calls from voice calls in 'trunk mode', you have to dial a fax prefix (e.g. *8 004944130005…) for each fax. This prefix (*8) is not a part of the called number will be deleted by the WLL 550 Pro.

    Default value: Phone and FAX at different interfaces
    Default value: not reported

- Timeout for autostart / Amount of digits for autostart in international order

    At POTS interfaces dialling can be made only digit by digit but a GSM call setup can be made en bloc only. Therefore the AS55X must collect all digits until the dialled number is complete.

    There are two ways to indicate the number to be complete:

    1st: Define a fix amount of digits for dialling to the GSM network. Set this amount to the maximum number of digits referring to your destination numbers.

    2nd: Define a timeout for waiting for the next digit. After this timeout expires, the call-setup to the GSM network will be started immediately.

    At the menu item "Timeout for autostart", the timeout for waiting for the next digit can be chosen.

    Default value: 4.0 seconds

    At the menu item "Amount of digits for autostart in international order", the fix amount of digits can be chosen. Select the maximum length of the called numbers in international form. E.g. for the longest dial number +491234567890 select 12 digits. If you are not sure about the maximum lengths of the number set this value to higher than 15.

    Default value: 21

5.5.2 Fax settings

This fax gateway is working in a buffering mode for optimal performance via the GSM networks that means for highest probability of transmitting a fax document successfully But in this mode, the sender of a fax does not get an acknowledge, if the fax document really reaches the receiver.

If the sender needs a confirmation whether the fax transmission has been successful or not an acknowledgement can be defined. The sender gets an error feedback when the transmission of the fax failed or a positive receipt when it has been successful. The acknowledgement will be generated to the sending fax device by the WLL 550 Pro.

In this environment, the far fax is that one, connected via the GSM network.

For more details about fax mode see 1.1.

- Positive acknowledge text to far FAX / Error feedback text to far FAX
- Positive acknowledge text to near FAX / Error feedback text to near FAX

    Enter any text to enable the feature. This text will be forwarded as positive acknowledgement resp. error feedback.

    For the positive acknowledgement and for the error feedback it is also possible to insert the destination number. Therefore use the "Insert destination number" button at the bottom of the window.

5.5.3 CLIR with FAX calls

Outgoing FAX calls present the voice number instead of the FAX number of the SIM. This can result into confusion by the recipient, which number should be used for a reply.

If unnamed bit 74 (*Special settings*) is set, the presentation of the number will be restricted, even if it is requested for voice calls. *> needs FW 3.16 <*

5.6   ISDN access

✋	ISDN access is only available for AS55X BRI and AS55X PRI devices.

5.6.1   General

- Amount of dialled digits to be ignored

  In some cases a number is assigned to the tie-line or cross connection the AS55X is connected to and this number is sent as a prefix to the dialled digits. In such cases, it is necessary to ignore the digits of this prefix.

  At this point, the amount of digits to be ignored can be selected.

  Default value: 0

- Timeout for autostart / Amount of digits for autostart in international order

  An ISDN dialling can be made either en bloc or digit by digit but a GSM call setup can be made en bloc only. Therefore the AS55X must collect all digits until the dialled number is complete.

  There are four possibilities to indicate the number to be complete:

  1st: The ISDN network indicates the completion with a special information element. This often happens by redialling.

  2nd: The calling subscriber indicates the number to be complete with typing a rhombus (#). This is rarely used in the field, because it is complicated and sometimes the PBX has problems with this digit.

  3rd: A fix amount of digits for dialling to the GSM network can be used. Set this amount to the maximum number of digits referring to your destination numbers.

  4th: A timeout for waiting for the next dialling digit can be defined. If this timeout expires, the call-setup to the GSM network will be started immediately.

  At the menu item "Timeout for autostart", the timeout for waiting for the next digit can be chosen.

  Default value: 4.0 seconds

  At the menu item "Amount of digits for autostart in international order", the fix amount of digits can be chosen. Select the maximum length of the called GSM number in international form. E.g. for the longest GSM number +491234567890 select 12 digits. This setting does not depend on the really dialled number format in ISDN. En bloc transferred numbers can be longer than this value.

  Default value: 11

- Change type of dialling number to… / Amount of CLIP digits from ISDN to be ignored / Change type of CLIP digits from ISDN / Type of CLIP to ISDN

  In ISDN domain, every dialled or dialling number additionally gets a type of number. Three types of number are in use (examples with country code "CC", network prefix "NNN", subscriber number "SSSSS", national prefix "0" and international prefix "00"):

  1st: Unknown number type with all prefixes
  (digits example: 0NNNSSSS or 00CCNNNSSSS)

  2nd: National number type without national prefix
  (digits example: NNNSSSS)

  3rd: International number type without international prefix
  (digits example: CCNNNSSSSS)

  o Change type of dialling number to…

  In some cases the ISDN PBX sends a type of number that does not fit to the dialled digits. E.g. it assigns the national number type to digits with the national prefix. With this setting, the ISDN number type can be overwritten by a suitable type.

  Default value: *No change*

  o Amount of CLIP digits from ISDN to be ignored

  Sometimes the ISDN PBX sends the number of the calling subscriber (CLIP) in an order or a number type, which is not accepted for a callback to the PBX.

  With this setting, an amount of CLIP digits can be ignored in order to get a number for a callback that will be accepted by the PBX.

  Default value: *0*

  o Change type of CLIP digits from ISDN

  Sometimes the ISDN PBX sends the number of the calling subscriber (CLIP) in an order or a number type, which is not accepted for a callback to the PBX.

  With this setting, the type of CLIP digits can be changed in order to get a number for a callback that will be accepted by the PBX.

  Default value: *No change*

  o Type of CLIP to ISDN:

With this setting, the number type of the CLIP from GSM presented to ISDN can be selected. The unknown types can get a leading "0" as some PBXs need this in order to handle callback properly.

Default value: Unknown type

- Prefix for suffix dialling to ISDN:

    For call-setups from GSM to ISDN, a prefix can be set here. E.g. a fixed part of a number block. By default, the ISDN number type is unknown. For a national or international number type, the numerical digits can be lead by a "N" or "I" character.

    Default value: No entry

5.6.2   MSN-GSM link list

✋	The MSN-GSM link list is only available for the AS55X BRI devices.

If you use your AS55X as ISDN slave (point-to-multipoint) you have no possibility to dial numbers directly to GSM. Therefore you can link a GSM number to each ISDN MSN. The Length of the list is limited to 100 entries. If the AS55X reach one of the linked MSNs a call to the linked GSM number will be initiated immediately.

5.6.3   Charge

- Type of charge information:

    The AS55X can generate time related charge information during calls from ISDN to GSM.

    The accumulation of charge steps starts with the GSM connection even if the ISDN protocol option "immediately connect" is chosen.

    If the interim type is selected, the charging information will be sent to the PBX with every step (AOC-D, advice of charge during the call). If "By end of call" is chosen, the steps will be accumulated and the information will be transferred to the PBX with the call termination (AOC-E).

    Default value: Off

- Step distance of:

    Select the step distance of charge information at this point. The first step is noted with the GSM connection.

    Example: If a step distance of 10 seconds is set here and a call from ISDN to GSM is made with GSM connection duration of 23 seconds, 3 steps will be accumulated.

    Default value: 15 seconds

5.6.4   Causes

For most of the ISDN events the use of cause messages are of common value. But some PBX needs different information for an error free call-setup.

- ... the called number is not valid

    This abort cause will be sent to ISDN if the dialled number is not valid. This case will occur, if the AS55X failed converting the dialled number from an ISDN number type to a valid GSM number type.

    Default value: Cause 28

- ...the call-setup should be rejected in case of presence of the CLIP from ISDN

    If the call to GSM shall be blocked by the AS55X in the case of presence of the CLIP from ISDN, the desired abort cause must be entered here.

    Default value: No entry, function off

- ... the requested GSM channel is busy or not registered at the network or the signal quality is too bad

    If the GSM channel, requested by the routing function, is not available, this abort cause will be sent to ISDN. This case will occur, if no member of the selected GSM channel group is registered to a GSM network with sufficient signal quality.

    Default value: Cause 34

- ... the call-setup failed before alerting

    This can occur, if the called number does not exist, if the called phone is switched off, if the GSM network is overloaded etc.

    Default value: Cause 17

- ... the call-setup failed before connection:

    This can occur, if the called mobile phone has bad RF contact to the GSM network or if the called subscriber rejects the call.

    Default value: Cause 17

- ... if the call-setup is aborted by busy tone:

    Some GSM networks do not terminate the call immediately if the called phone is busy, they only switch a busy tone to the calling phone. If the AS55X should detect this busy tone and terminate the call-setup immediately, the desired ISDN cause must be typed in here.

    Default value: No entry, detection off

- ... if the call is aborted by announcement

  Some GSM networks do not terminate the call immediately if the called phone is not available, they only switch an announcement to the calling phone. If the AS55X is wanted to detect this announcement and terminate the call-setup immediately, the desired ISDN abort cause must be typed in here.

  Default value: No entry, detection off

- ... if the connection is aborted by busy tone via GSM

  Some GSM networks do not terminate the call immediately after the GSM subscriber has hang up, they only switch a busy tone to the calling phone. If the AS55X is wanted to detect this busy tone and terminate the call immediately, the desired ISDN abort cause must be typed in here.

  Default value: No entry, detection off

- ... the connection is cleared by the GSM subscriber

  This cause will be sent to ISDN with the clearing of the connection by the GSM phone.

  Default value: Cause 16

- List of ISDN causes

Cause 1	Unallocated (unassigned) number
Cause 2	No route to specified transit network
Cause 3	No route to destination
Cause 6	Channel unacceptable
Cause 7	Call awarded and being delivered in an established channel
Cause 16	Normal call clearing
Cause 17	User busy
Cause 18	No user responding
Cause 19	No answer from user (user alerted)
Cause 21	Call rejected
Cause 22	Number changed
Cause 26	Non-selected user clearing
Cause 27	Destination out of order
Cause 28	Invalid number format
Cause 29	Facility rejected
Cause 30	Response to STATUS ENQIRY
Cause 31	Normal, unspecified
Cause 34	No channel available
Cause 38	Network out of order
Cause 41	Temporary failure
Cause 42	Switching equipment congestion
Cause 43	Access information discarded
Cause 44	Requested channel not available
Cause 47	Resources unavailable, unspecified
Cause 49	Quality of service unavailable
Cause 50	Requested facility not subscribed
Cause 57	Bearer capability not authorized
Cause 58	Bearer capability not presently available
Cause 63	Service or option not available, unspecified
Cause 65	Bearer capability not implemented
Cause 66	Channel type not implemented
Cause 69	Requested facility not implemented
Cause 70	Only restricted digital information bearer is available
Cause 79	Service or option not implemented, unspecified
Cause 81	Invalid call reference value
Cause 82	Identified channel does not exist
Cause 83	A suspended call exists, but this call identity does not
Cause 84	Call identity in use
Cause 85	No call suspended
Cause 86	having the requested call identity has been cleared Call
Cause 88	Incompatible destination
Cause 91	Invalid transit network selection
Cause 95	Invalid message, unspecified
Cause 96	Mandatory information element missing
Cause 97	Message type non-existent or not implemented
Cause 98	Message incompatible or message type non-existent or not implemented
Cause 99	Information element non-existent or not implemented
Cause 100	Invalid information element contents
Cause 101	Message not compatible with call state
Cause 102	Recovery on timer expiry
Cause 111	Protocol error, unspecified
Cause 127	Interworking, unspecified

5.7 SIP access

⚠️	Notice: If you use the AS55X in SIP mode, the ISDN interfaces will be deactivated. If you use the WLL550 Pro in SIP mode, the analogue interfaces will be deactivated.

✋	SIP access is not available in all SMS/data only devices.

5.7.1 General

- SIP operation mode

  o Off

  No SIP abilities available, AS55X with BRI or PRI interface operates in ISDN mode.

  o SIP Server:

  One or more SIP clients can register to the AS55X as server. With the registration, the AS55X collects the connection parameters of the clients, their phone numbers, IP addresses and port numbers in a table. Based on these parameters call-setups from GSM to SIP can be routed to the destination by the phone number.

  An overview off all registered users can be found in the "view SIP registered user list" in "services" menu (see 7.9).

  To control the access of clients to the SIP server, you can use the "authorized user list" (see 5.7.4). In this case, only entered users can register to the AS55X.

⚠️	Notice: It is recommended that the AS55X operates with fixed IP addresses instead of DHCP assigned IP addresses

  o SIP Client:

  The AS55X registers to a SIP server within or outside of the LAN. All call-setups from GSM to SIP are routed via this server. In most environments the operation mode "SIP client" is not recommended. Call-setups to GSM are very complicated and normally it must be worked with DTMF (telephone event) suffix dialling. In some cases you can use the "GSM number dialling" (see 5.7.2).

  This mode is mandatory if you want to use GSM-PBX mode of the AS55X.

  The status of your client connection can be checked in "services"-menu in column "view SIP status" (see 7.8).

  o SIP peer mode

Generally the AS55X is connected (via LAN) to a PBX. In this case the SIP "peer" mode is the best choice. The AS55X is directly linked to the PBX. It only accepts calls from peer IP address and it routes all call-setups from GSM to the peer IP address. Optional the PBX can register to the AS55X.

You can configure two peers, but the AS55X will always proceed to contact the first peer. If this is not reachable the AS55X will contact the second peer. Incoming call are processed by both peers.

⚠️	Notice: It is recommended that the AS55X operate with fixed IP addresses and not with DHCP assigned IP addresses.

- Amount of dialled digits from SIP to be ignored

    Some PBXs sent the dialled number with a prefix to the wired network. The amount of prefix digits can be ignored by AS55X to ensure a correct routing.

    The amount of digits to be ignored, can be selected here.

    Default value: *0*

- Prefix for suffix dialling to GSM

    Some PBXs sent the dialled number without a leading zero or with another incorrect string. To correct the prefix you can fill in a suffix that is assigned to the GSM network.

    Default value: *not reported*

- Amount of CLIP digits from SIP be ignored

    If you use the callback function of the AS55X, you can make an adjustment of the CLIP here. When the PBX sends more CLIP digits of a call than needed for a callback, you can set the amount of digits that have to be ignored.

    You can verify the correct function in the "call setup list" in services menu (see 7.5).

    Default value: *0*

- Prefix for CLIP from SIP

    If you use the callback function of the AS55X, an adjustment of the CLIP might be necessary. When you need some additional digits for a correct routing, you can fill in up to 21 digits in front of the saved number.

You can verify the correct function in the "call setup list" in services menu (see 7.5).

Default value: *not reported*

- Type of CLIP to SIP

    This parameter is only effective when the AS55X is working as SIP-Server or as SIP Peer. In this case the shown CLIP can be set here. You have five different values:

Unknown	030 123456
Unknown with prefix "0"	0 030 123456
Unknown with country code	0049 30 123456
Unknown with country code and prefix "0"	0 0049 30 123456
International type with "+"	+49 30 123456

    In case of SIP-Client the AS55X normally uses the SIP username for CLIP. Otherwise you can use the function "GSM CLIP in client mode" (see 5.7.2).

    Default value: *Unknown type*

- Prefix for suffix dialling to SIP

    Referring to "Prefix for suffix dialling to GSM" you can define a prefix for dialling to SIP. If your PBX expects a string with main and extension number, you can fill in the main number for example and the user only needs to dial the extension number.

    Default value: *not reported*

- Maximum amount of digits for suffix dialling to SIP

    For call-setups from GSM to SIP, the amount of DTMF suffix digits should be set, if the length is fixed. This allows the AS55X a fast detection of a complete number. If the length is not fixed, this setting should correspondent to the maximum length.

    Default value":   *11*

- Detect suffix dialling to SIP complete after… / … or with DTMF digit

    Referring to the maximum amount of digits for suffix dialling to SIP, the AS55X checks if the length is reached, otherwise it detects the dialled number as complete after a chosen duration or after the chosen DTMF digit.

    Default value: *4 seconds / #*

5.7.2   Extended (expert mode)

> ⚠️ Notice: To view this menu you have to start the AS55X service software in expert mode by opening the tab "Preferences" in the menu bar.

- Protocol option immediate connect simulation

  Depending on the PBXs properties, sometimes announcements and tones from the GSM network during the call-setup are not switched to the calling user. With this protocol option, the AS55X simulates an active call when the call setup to the GSM network starts. The PBX will handle the call like an active one and transfer audio to the caller.

  Default value: *Off*

  > ⚠️ Notice: If there is a connection with costs between AS55X and PBX, charges for this line will occur already during the call-setup phase.

- Protocol option supervise connection <u>by sip info</u>

  There can be a lot of problems or reasons for a termination during a call. To avoid that there is a noteless break in the IP connection you can use this supervise function.

  If a SIP call is in active mode, the AS55X can actively supervise the connection via SIP reINVITE requests or SIP INFO messages. So every 60 seconds a message will be sent by the AS55X and the opposition will answer. If there is no reply, the AS55X will assume that there is a termination and will release all connection parameter. The connection will be aborted exactly after one minute.

  This is recommended in all cases. If it does not work, try change between SIP INFO and SIP reINVITE. If both methods are not supported by the PBX, you have to switch off this supervision.

  Default value: *… by reINVITE*

- Incomplete SIP INVITE handling

  Standard SIP does not know procedures like overlap sending/receiving in ISDN, but there are some efforts, to implement such functions.

  If the incomplete SIP INVITE handling is off, every received INVITE will be interpreted as complete, this is the normal case. With all other settings, an INVITE is detected as complete, if either the amount of digits for autostart is included or the timeout for autostart has expired or the dialled number matches an entry of the individual call-setup list under wired access. The handling of the other cases can be set here.

  o *Keep INVITE unanswered:*
  No response is made to an incomplete INVITE.

o *Respond TRYING to first INVITE:*
Only the first incomplete INVITE will get TRYING as response, all further INVITEs are not answered.

o *Respond TRYING to every INVITE:*
Every incomplete INVITE will get TRYING as response.

o *Respond ADDRESS INCOMPLETE to first INVITE:*
Only the first incomplete INVITE will be rejected with the final result code ADDRESS INCOMPLETE.

o *Respond ADDRESS INCOMPLETE to every INVITE:*
Incomplete INVITES are rejected with the final result code ADDRESS INCOMPLETE.

In normal cases, the PBX is collecting dialled digits until they are complete or until the user marks the number as complete. In these cases the incomplete INVITE handling should be off.

Default value: Off

- Timeout for autostart

    After this period of no further INVITE, an INVITE will be handled as complete. Refer to incomplete SIP INVITE handling.

    Default value: 4 seconds

- Amount of digits for autostart in international order

    With this amount of dialled digits, an INVITE will be handled as complete. Refer to the incomplete SIP INVITE handling. Please keep in mind, the number is converted to international type before comparison.

    Default value: 11

- GSM CLIP in Client mode

    Normally a SIP server only accepts call-setups with the same local username as in the registration, hence by default the AS55X always uses the configured local username for this case and never passes the GSM CLIP. But if the server can make sense of the GSM CLIP, it can be inserted as well.

    If your server supports GSM CLIP you can variegate between durable GSM CLIP or an option with a comparing to the GSM-PBX link list. If there is a match, the linked

number will be transmitted. Otherwise you will see the GSM CLIP link in "always pass GSM CLIP" mode. Alternative you can select that the GSM CLIP is passed only if there is no match in the GSM PBX link list.

Default value: *Never pass GSM CLIP*

- Request SIP PRACK sequence

  In cases of missing early media or in cases of one way audio it is possible to request or suppress this sequence unconditionally.

  If the AS55X is working with the value "never" there are no PRACK requests at any time. If you use "always" every SESSION PROGRESS requests a PRACK. With the setting "if supported" the AS55X evaluate the received INVITE message. If there is a "100REL" offer within a supported header, the AS55X uses PRACK requests, otherwise the requests are not longer required.

  Default value: *If supported*

- Allow direct GSM number dialling in client mode

  Normally in SIP client mode only DTMF suffix dialling is possible. But sometimes it´s also possible, that there are requirements for complete number dialling in client mode. Therefore you can check this box.

  The AS55X will work like SIP client and SIP peer together. All received numbers, except the configured local username and the SIP numbers of the GSM PBX link list, will cause a direct GSM call. These numbers could be received from anywhere. Therefore also check the next configuration box (Require proxy) to avoid illegal usage.

- Require proxy / Only accept call-setup from peer(s)

  This checkbox has different names in SIP peer and in SIP client mode.

  In SIP client mode you can configure direct GSM number dialling if it is possible (see configuration checkbox above). Normally everybody could send a call-setup. To avoid unauthorised usage, the AS55X uses the SIP contact header for restricting this access. Only the SIP Server is legitimate to sent messages, because on SIP client registration the CONTACT header will be forwarded. So it is only possible, that the SIP Server knows the correct header. If your SIP Server does not support this feature, you can restrict this access only to the IP address of the SIP registrar. Therefore you have to set bit 52 in special settings (see 0).

By default, in SIP peer mode the AS55X only accepts SIP calls from the connected peers. If you want to receive numbers from other clients you can uncheck this box. Then call-setups are not only accepted from the two peer IP addresses.

Default value: *checked / checked*

- Transport protocol

  Normally SIP is using UDP. But some provider or manufactures are using SIP over TCP. Therefore you can change the transport protocol. If you choose TCP, SIP messages are only accepted with TCP. UDP messages will be ignored. If you choose UDP, it is vice versa.

  The transmission of RTP data will be done exclusively with UDP.

  Default value: *UDP*

- VLAN ID / VLAN priority

  If you use VLAN in your local network, you can set a VLAN ID here. This is used to assign to a special VLAN on a VLAN trunking or VLAN tagging port. Only SIP and RTP data will be tagged with this VLAN ID. All other VLAN data will persist untagged.

  If you enter no VLAN ID, the AS55X works without VLAN. The VLAN ID value is allowed between 1 and 4095 (12 bit). The VLAN Priority, which you can set in the next box, can be set between 1 and 7 (3 bit).

  Default value: *not reported / not reported*

5.7.3   Names and passwords

- Local hostname

  Here you can define the hostname of your device. This name is used in the message header in the SIP protocol. If you define no name, the IP address is set as part of the SIP-URI.

  If you use the AS55X as SIP client, you are part of the SIP domain of your provider. Therefore it´s recommended to fill in the name of the remote host.

  If you use the AS55X in server mode it is mandatory to set the local hostname.

  Default value: *not reported*

- Remote hostname / Registrar IP address

If you use SIP client mode, you have to configure the address of your SIP Server. Therefore you can enter the remote hostname or the registrar IP address. For remote hostname you have to configure the IP address of the DNS server. Then the hostname will be translated into the IP address. The translated or the directly entered Server IP is listed in "view SIP status" in services menu (see 7.8).

If you entered both, remote hostname and IP address, the hostname will not be translated for addressing of the SIP packages. It is only used for the request-line and for the TO header in the SIP message. The destination of the packages is determined by the IP address.

In SIP peer mode it is nearly the same. You have to enter the peer IP address into the peer IP address field. All packages will be forwarded to this IP address. If you additionally enter the remote hostname, the name is transmitted in the request line and in the TO header.

This function is greyed out in SIP Server mode.

Default value: *not reported / not reported*

- Local username / Authenticate ID

    In SIP client mode, the local username and sometimes also the authentication ID is required for the registration to the SIP server. Mostly you have to fill in your telephone number. If you don´t have an additional authentication ID, keep the field blank. The username, or if the ID is entered, this one, is used for the REGISTER in FROM header, in TO Header, in CONTACT header and also in AUTHORISATION header If the AS55X sends an INVITE, the local username is used for FROM header and for the CONTACT header. In the AUTHORISATION header the authentication ID is used.

    If you don´t enter the authentication ID, all fields in the SIP header will be used with the local username.

    This entry is case sensitive.

    This function is greyed out in SIP server and in SIP peer mode.

    Default value: *not reported / not reported*

- Authenticate Password

    Here you have to fill in your password for authentication at your SIP server.

    This function is greyed out in SIP server and in SIP peer mode.

    This entry is case sensitive.

    Default value: *not reported*

- Local SIP port

    Here you can change the SIP Port of the AS55X.

    Default value: *5060*

- Remote SIP Port

    This is the SIP port number of the PBX. The standard value is 5060, but sometimes another port number is used. If it is unknown, in peer mode a look into the "SIP registered user list" (see 7.9) can help if the PBX has already registered itself.

    This function is greyed out in SIP server mode.

    Default value: *5060*

- Peer IP address / 2nd peer IP address

    These IP addresses have two functions. On the one hand the AS55X sends call setups only to the PBXs on these IP addresses. On the other hand call setups to GSM are accepted only from these IP addresses. The access is locked to the dedicated PBXs. Keep the "only accept call-setup from peer(s)" function in mind (see 5.7.2).

    All call setups to the PBXs are routed to the (1st) peer IP address except if the connection to this address is not given but the connection to the 2nd peer IP address is present.

    The connection between AS55X and your PBX will be checked continuously by SIP OPTION messages every 10 seconds.

    This function is greyed out in SIP server and in SIP client mode.

    Default value: *not reported*

5.7.4   Authorized user list

If the access to this gateway has to be restricted to a group of up to 100 users, fill in the connection data of the members here.

If this list is completely empty, there is no restriction and all subscriber can use the AS55X for outgoing calls to GSM. If an entry consists of a username without password, no authentication is necessary for this user. If a password is defined, the AS55X generates an authentication request to the user.

Enter numeric digits for the user number, optionally leaded by a "+" character.

It is possible to use this list with wildcards. Therefore you can enter a user number with less numerics than your phone number (e.g. enter 10; for the users 1000 up to 1099). All users can register with the same username, if entered, also with the same password.

Default value: *not reported*

5.7.5   Responses

- ... the called number is incomplete

  This reason will be sent to SIP, if the INVITE has been detected as incomplete. Refer to the helpline of incomplete SIP INVITE handling.

  Default value: *484 (address incomplete)*

- ... the called number is not valid

  This abort cause will be sent to SIP if the dialled number is not valid. This case will occur, if the AS55X failed. This case will occur, if the AS55X failed converting the dialled number from a SIP number type to a valid GSM number type.

  Default value: *484 (address incomplete)*

- ... the call-setup should be rejected in case of presence of the CLIP via SIP

  If the call shall be blocked by the AS55X in the case of presence of the CLIP from SIP, the desired abort reason must be entered here.

  Default value: *not reported*

- ... the requested GSM channel is busy or not registered at the network or the signal quality is too bad

  If the GSM channel, requested by the routing function, is not available, this abort cause will be sent via SIP. This case will occur, if no member of the selected GSM channel group is registered to a GSM network with sufficient signal quality.

  Default value: *503 (Service unavailable)*

- ... the call-setup failed before alerting

  This can occur, if the called number does not exist, if the called phone is switched off or if the GSM network is overloaded.

  Default value: *486 (busy here)*

- ... the call-setup failed before connection

This can occur, if the called mobile phone has bad radio contact to the GSM network or if the called subscriber rejects the call.

Default value: *486 (busy here)*

- ... if the call-setup is aborted by busy tone

  Some GSM networks do not terminate the call immediately if the called phone is busy, they only switch a busy tone to the calling phone. If the AS55X is wanted to detect this busy tone and terminate the call-setup immediately, the wanted SIP abort reason must be typed in here.

  Default value: *not reported*

- ... if the call-setup is aborted by announcement

  Some GSM networks do not terminate the call immediately if the called phone is not available, they only switch an announcement to the calling phone. If the AS55X is wanted to detect this announcement and terminate the call-setup immediately, the wanted SIP abort reason must be typed in here.

  Default value: *not reported*

- ... if the connection is aborted by busy tone via GSM

  Some GSM networks do not terminate the call immediately after the GSM subscriber has hung up, they only switch a busy tone to the calling phone. If the AS55X is wanted to detect this busy tone and terminate the call immediately, the wanted SIP abort reason must be typed in here.

  Default value: *not reported*

- List of SIP responses

Response 400	Bad request
Response 401	Unauthorized
Response 402	Payment required
Response 403	Forbidden
Response 404	Not found
Response 405	Method not allowed
Response 406	Not acceptable
Response 407	Proxy authentication required
Response 408	Request timeout
Response 409	Conflict
Response 410	Gone
Response 411	Length required
Response 413	Request entity too long
Response 414	Request URI too long
Response 415	Unsupported media type
Response 420	Bad extension
Response 480	Temporarily unavailable
Response 481	Call leg does not exist
Response 482	Loop detected
Response 483	Too many hops
Response 484	Address incomplete
Response 485	Ambiguous
Response 486	Busy here
Response 488	Not acceptable here
Response 500	Server internal error
Response 501	Not implemented
Response 502	Bad gateway
Response 503	Service unavailable
Response 504	Server timeout
Response 505	Version not supported
Response 600	Busy anywhere
Response 603	Decline
Response 604	Does not exist anywhere
Response 606	Not acceptable

5.7.6   DTMF processing

- Convert telephone-event from LAN (RFC2833/4733) to DTMF tones to GSM

  During an active call, telephone events according to RFC2833/4733 from the remote subscriber will be converted to DTMF tones that are sent to GSM e.g. for controlling mailboxes.

  Default value: unchecked

- Convert DTMF-relay from LAN to DTMF tones to GSM

  During an active call, SIP INFO messages within submitted DTMF events from the remote subscriber will be converted to DTMF tones that are sent to GSM e.g. for controlling mailboxes.

  Default value: unchecked

- Detect restart sequence (*0) during call setup from GSM

  If this box is checked, for the call setup direction from GSM to PBX, the GSM caller can abort the current call setup to the PBX with the DTMF tones " * " and " 0 " immediately. He will hear the dial tone again and can start a new trial.

  This function cannot work, if the call setup to the PBX is made via the subscriber number without delay.

  Default value: checked

- Detect restart sequence (*0) from GSM during active call

  During an active call, the GSM subscriber can abort the call by typing the DTMF tones " * " followed by " 0 " instantly. He will hear the dial tone and can start a new call setup to the PBX.

  This function cannot be mixed with converting DTMF tones from GSM to telephone-events to LAN.

  Default value: checked

- Convert DTMF tones from GSM to telephone-event to LAN (RFC2833/4733)

  During an active call, DTMF tones from GSM will be converted to telephone events according to RFC2833/4733 that are sent to the PBX e.g. for controlling a mobile client at the PBX.

This function cannot be mixed with detecting the restart sequence during an active call.

Default value: *unchecked*

⚠️	Keep in mind that no " * " telephone-event will be transmitted if you use GME function. Also another event will be suppressed for 2 seconds after pressing the " * ".
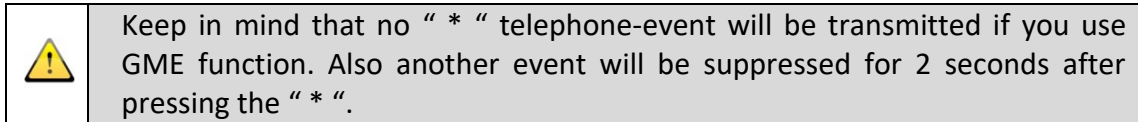
- Convert DTMF tones from GSM to DTMF-relay to LAN

    During an active call, DTMF tones from GSM will be converted to DTMF events which will be submitted via SIP INFO message e.g. for controlling a mobile client at the PBX.

    This function cannot be mixed with detecting the restart sequence during an active call.

    Default value: *unchecked*

⚠️	Keep in mind that no " * " telephone-event will be transmitted if you use GME function. Also another event will be suppressed for 2 seconds after pressing the " * ".
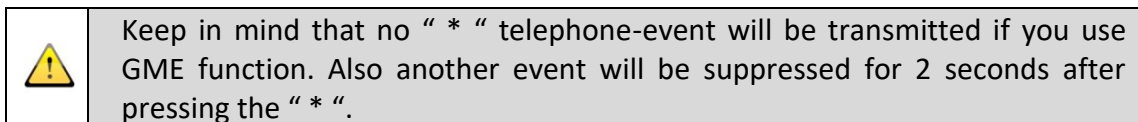
5.7.7   Codecs

- First codec / Second codec

    Set the codec types that are offered in a call setup to the PBX and are accepted in a call setup from the PBX. The codec choice is made in the order entered here.

    Normally all codecs should be allowed and the order is not important. But for GME operation with call transfer scenarios a restriction may be useful.

    If the PBX only offers not supported codecs, the gateway will answer with SIP response 415 (UNSUPPORTED MEDIA TYP) and no call will be initiated. For incoming calls the AS55X will choose the first applicable codec without checking the priority.

    Default value: *G711 A-law / PCMA*
                                              G711 U-law / PCMU

5.8   Wired access

✋	"Wired Access" is not available in all SMS/data only gateways.

5.8.1   Country specific settings

- Country specific setting for…

  Here you can choose one of some predefined areas. Then the following fields with national prefix, international prefix and area code will be filled automatically.

  Default value: *unknown but required*

- National prefix

  In most cases the international prefix is "0", but it varies from country to country.

  For properly number converting from ISDN to GSM domain and vice versa type in the valid national prefix of the ISDN site nationality.

  Default value: *0*

- International prefix

  In most cases the international prefix in wired networks is "00", but it varies from country to country. In GSM networks the general international area code is the plus digit ("+").

  For properly number converting from ISDN to GSM domain and vice versa type in the international prefix of the ISDN site nationality.

  Default value: *00*

- Country Code

  For properly number converting from ISDN to GSM domain and vice versa type in the country code of the area this gateway is located.

  Default value: *not reported but required*

5.8.2   Global call-setup

For more details about callback mode see 1.1.

In the following you can draw some distinctions for callback mode like saving duration or about the types of saved calls.

For saving the calls in the call setup list and for a possible recall both sides have to display their CLIP.

For a faultless function you also have to activate the "callback via call-setup list" for every GSM channel (see 5.10.2).

To check the faultless function all entries in the call setup list are shown in the "Services" tab (see 7.5).

- Save call setup type

  Here you can distinguish which data will be saved in the call setup list. You can define that every call will be saved ("Every call") or that only an unanswered call will be saved. This means that the call setup will only be saved if the called party did not answer. Keep in mind if the call is accepted by a mailbox, the call-setup will not be listed. The same may apply to network announcements.

  Default value: *Off*

  > ⚠ Notice: The correspondent setting in the window "Wired access" / "Individual call-setup" will overrun this one.

- Save call setup duration

  Here you can choose the duration of the storage. It is possible to select a period between 15 minutes and one week. After this period, the call-setup will be deleted.

  Default value: *1 hour*

- Save call setup event

  With this choice the specific event can be defined, that initiates the saving of the call-setup data set. Notice: If other selections than "By reception of idle tone from GSM" are made, it is not guaranteed that the called subscriber has realized the called subscriber has realized the call attempt.

  Default value: *By reception of idle tone from GSM*

- Delete entry after successful callback

  If the entry in the call-setup list should be deleted after the recall, then the checkbox should be ticked.

  Default value: *ticked*

- Save Call Setup only from PBX Domain

  If this box is unchecked, calls will be saved only if the source of the call is in PBX domain. Calls from the fixed network, that are forwarded (by an extension) to a GSM phone will not be saved in this case.

  Default value: *ticked*

- Global routing

  You have the option to create an external routing for every mobile number. Therefore you can use the individual call-setup menu below. Each of the entered number can be forwarded to an individual group of GSM channels. Only those groups can be selected, that are defined in the window "GSM settings" / "GSM groups" (see 5.9.3) before. Here you can define the default, global route. Every number which is not listed in individual call-setup parameters will be routed as defined. If "no call-setup" is selected, every call-setup attempt to an unspecified number will be rejected by the AS55X. If "Next ready GSM channel" is selected, every call-setup attempt to an unspecified number will be routed via the next channel, that is registered, that has sufficient signal quality and is not in use.

  Default value: *Next ready GSM channel*

⚠	Notice: The correspondent setting in the window "Wired access" / "Individual call-setup" will have priority over this one.

- Use individual call setup parameters

  With this check box, the use of the list of individual call-setup parameters can be enabled or disabled. If it is disabled, the global routing and the general GSM settings will be valid for every call-setup from ISDN to GSM. If it is enabled, the entries in the list of individual call-setup parameters have priority over the global routing and GSM general settings.

  Default value: *ticked*

5.8.3   Individual call-setup

In the individual call setup list, you can enter up to 5000 values.

- (Beginning of) GSM number

    If you enter a complete GSM subscriber number, this data will be valid for just one GSM subscriber. By leaving the last digits of the GSM subscriber number blank, this data will be valid for a group of GSM subscribers, whose numbers have these first digits (part of number, e.g. network prefix or VPN prefix) in common .

    A longer number has a higher priority, so you can make exceptions within the group.

    The number has to be entered in international order, beginning with "+". A VPN number must be entered with the VPN prefix.

    All settings that are made to this individual number will have priority over the global or general settings.

    This numbers are also used for call interception in PassThrough mode (see 5.11.1).

- (Beginning of) PBX number

    To exclude single subscriber numbers from the routing via the AS55X or to define a specific behaviour depending on the calling party number you can add a PBX subscriber number to this list.

    The PBX subscriber number list has a higher priority than the GSM number list .

    The calling party number has to be entered as full number in national order or just as PBX extension number. Have a look in the "View number management" to see how the calling party number are transmitted by the PBX.

    All settings that are made to this individual number list will have priority over the global or general settings.

    This numbers are also used for call interception in PassThrough mode
    (also refer to 1.8).

- Maximum suffix digits

    If the GSM number is a complete one, type in "0" here. Otherwise define how many digits can follow at most.

For normal operation, this entry optimizes the call-setup speed. In Pass Through mode, it controls the switching of a call-setup to a GSM channel.

If this entry is not zero, the detection of a complete number will additionally be controlled by "Timeout for autostart" in the window "ISDN access" / "General".

Default value: *0*

- Routing

    Referring to the "Global call-setup" menu here it is possible to define a routing for every single number or groups of numbers. Therefore you have to create channel groups (see 5.9.3). Only these groups can be selected for the external routing. It is possible to select a particular routing for every line. If "no call-setup" is selected the line will be marked red and every call-setup attempt to an unspecified number will be rejected by the AS55X. If "Next ready GSM channel" is selected, every call-setup attempt to an unspecified number will be routed via the next channel, that is registered, that has sufficient signal quality and is not in use. If "Use global routing" is selected, the setting of the global routing in the window "ISDN access" / "Global call-setup" will take place (see 5.8.2).

    Default value: *Use global routing*

- Display GSM number

    Here you can choose whether your number will be transmitted (YES / CLIP) or if not (NO / CLIR). If you choose "Network settings", the default setting within the GSM network is valid. "As on wired side" means that the own SIM number is displayed if a CLIP from the wired side is present.

⚠️	Notice: The weighting of the CLIP settings is as follows. The lowest priority is the setting in general GSM settings (display number -> GSM Network), then in individual call setup, and the highest setting is the CLIP/CLIR-Code in .General GSM settings.

    Default value: *Network setting*

- Save call-setup type

    Referring to the "call setup type" in the "Global call-setup" menu (see 5.8.2) you can here distinguish which data will be saved in the call setup list for special, single numbers. You can define that every call will be saved ("Every call") or that only an unanswered call will be saved. This means that the call setup will only be saved if the called party does not answer. Keep in mind, if the call is accepted by a mailbox the call-setup will not be listed. The same may apply to network announcements. If you choose

For saving the calls in the call setup list and for a possible recall both sides have to present its CLIP.

To check the faultless function all entries in the call setup list are shown in the "Services" tab (see 7.5).

Default value: *Off*
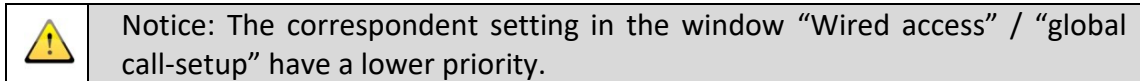
⚠️	Notice: The correspondent setting in the window "Wired access" / "global call-setup" have a lower priority.

- Save call setup duration

  Here you can choose the duration of the storage. It is possible to select a period between 15 minutes and one week. After this period, the call-setup will be deleted.

  Default value: *1 hour*

⚠️	Notice: The correspondent setting in the window "Wired Access" / "Global call setup" has a lower priority.
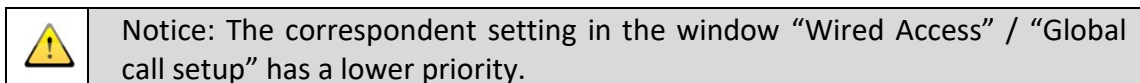
- Delete entry after callback

  If the entry in the call-setup list should be deleted after the recall, then select "yes", otherwise the entry persists after a successful recall.

  Default value: *not reported*

- Send SMS callback request

  For this specific number or group of numbers it can be defined, if a short message request should always or never be sent, This will override the global settings in "Short message callback request" (refer to 5.8.5).

  "Always" means, the short message will be sent even if the call was successful.

  Default value: *As globally defined*

5.8.4   Short PBX Domain number list

By default, the AS55X assumes a wired number to be 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 this list. That means, the AS 55X will treat all here defined number-beginnings as extension numbers of the PBX.

The numbers have to be entered in international order, beginning with '+'.

5.8.5   Short message callback request

For more details about "short message callback request" see 1.4.

- Message

  Here the short message text can be entered. The text is limited to 160 characters. In the right upper corner of this window you have a counter which shows the amount of entered characters. This text is always the same for all short messages and can be complemented by the right handed variables. To enter one variable press the belonging button. To delete this variable press this button again.

  Default value: *not reported*

- DTMF Code

  This code the caller can enter to send a short message manually. It can be entered while the called party is ringing, is busy or if the call has been rejected. After entering this code the call will be determined. You can also enter it during the first 10 seconds of an established call if the checkbox "Enter code during connection" is checked.

  If the checkbox "Send message automatically if call is unsuccessful" is not checked you will be invite to enter this code by selected tone if the call is finished by the called party (e.g. the called party is busy or reject the call).

  If you configured "Send message automatically if call is unsuccessful" and a mailbox answers the call, no short message will be created, because this call will be handled as successful. Therefore also use this DTMF code.

  Default value: *not reported*

- Enter code during connection

  Here you can allow to enter a DTMF code during the first ten seconds of a connection. This is necessary, if it should be possible to send a message after a voicemail has answered the call. If you configured "Send message automatically if call is unsuccessful" and a mailbox answers the call, no short message will be created, because this call will be handled as successful. Therefore also use this DTMF code.

  ⚠️ Notice: If GME is configured as well, the GME code cannot enter during the first ten seconds of the call.

  Default value: *unchecked*

- Send message automatically if call is unsuccessful

  If this box is checked, the message will be sent automatically if the caller hangs up while the destination is ringing, if the destination is not reachable, if it is busy or has rejected the call.

⚠	Notice: Some GSM networks do not terminate the call immediately if the called phone is busy, they only switch a busy tone to the calling phone. In order to detect the busy tone, please refer to "ISDN access – Causes" (see 5.6.4) or "SIP access – Responses" (see 5.7.5)

  Default value: *unchecked*

- DTMF code and DTMF number enter tone

  This tone or announcement you will hear if you have to enter the DTMF code or the DTMF number.

  Default value: *not reported*

5.9 GSM settings

5.9.1 General

- Equipped GSM channels

  The amount of GSM channels, installed in the AS55X has to be set here.

  Be default the amount is set of the maximum of available channels of your single devise. If you want to use an AS551 Slave or an AS55X with a smaller amount of channels you have to decrease or increase this value, because the AS55X do not detect the installed modules automatically. All excessive channels will be designated with errors.

  Default value: 2 (AS550) / 6 (AS551)

⚠	Notice: Please do not use this setting to temporarily switch off GSM channels for test purposes, because configurations to not installed channels will be cleared. For temporarily deactivation use "GSM settings / GSM channel" (see 5.9.2)

- CLIP code from PBX

  If you configure CLIR in the field below ("Display number -> GSM network"), it is possible to submit your number by entering a code in front of your dialled number. This code you have to enter here. It can consist of up to four numeric digits.

  Default value: not reported

⚠	Notice: The weighting of the CLIP settings is as follows. The lowest priority is the setting in general GSM settings (display number -> GSM Network), then in individual call setup, and the highest setting is the CLIP/CLIR-Code in .General GSM settings.

- CLIR code from PBX

  If you configure CLIP in the field below ("Display number -> GSM network), it is possible to suppress your visible number by entering a code in front of your dialled number. This code you have to enter here. It can consist of up to four numeric digits.

  Default value: not reported

⚠	Notice: The weighting of the CLIP settings is as follows. The lowest priority is the setting in general GSM settings (display number -> GSM Network), then in individual call setup, and the highest setting is the CLIP/CLIR-Code in general GSM settings.

- Display number -> GSM network

  Here, you can generally submit or suppress your caller ID (CLIP / CLIR).
  The setting "As on wired side" means that the own SIM number is displayed
  if a CLIP from the wired side is present.

  Default value: *Network setting*

⚠	Notice: The weighting of the CLIP settings is as follows. The lowest priority is the setting in general GSM settings (display number -> GSM Network), then in individual call setup, and the highest setting is the CLIP/CLIR-Code in general GSM settings.

- GSM channel restart timer

  In some cases it can happen that a logged-in module is no longer visible in the network. To prevent this case, you can restart each GSM module.

  So the module log in again after the chosen period. If you select "Off", the module will not be restarted.

  This is the global default value for all channels. For a precision tuning to each channel you can use "GSM settings / GSM channels".

  Default value: *after 8 hours*

⚠	Notice: A restart will never be triggered while a GSM connection or a call-setup is in progress.
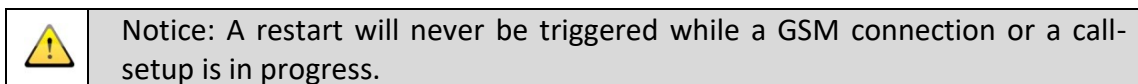
- GSM channel restart mode

  In some cases it happens that a logged-in module is no longer visible in the network. To prevent this case, you can restart each GSM module.

  Here you can select the global setting referring to the restart mode of the GSM channels. You can choose "without GSM connection". Then the AS55X will restart after the last call or call setup. If you choose "unconditional" the AS55X will restart definitively after the chosen period.

  Default value: *Without GSM connection*

⚠	Notice: A restart will never be triggered while a GSM connection or a call-setup is in progress.
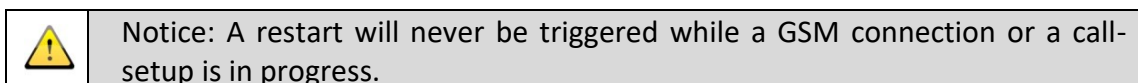
- Timeout DTMF recognition

    A DTMF tone, sent by a mobile phone often has a length of about 300 to 600 milliseconds. It might occur that a tone is divided into two parts by a short disturbance of GSM reception. That might cause a faulty dialling to the ISDN domain.

    To prevent that the interface recognizes the same tone twice, a timeout can be set. Within this timeout the same DTMF tone is not accepted.

    Default value: *300 milliseconds*

5.9.2   GSM channels

- On/Off

    Every single channel can be switched on or off. If it is switched off, all settings to this GSM channel in this and other windows will be marked red. The indicator of a disabled GSM channel permanently shows red.

    This setting should be used, if the channel should permanently or temporary (e.g. for test purposes) be disabled. Not additional setting to this channel will be affected.

    Default value: *On*

- Net code

    To force a registration to an explicit GSM network, the net code of the desired GSM network can be set at this point. An individual net code can be assigned to each GSM channel. This function will be activated if a number is typed in.

    The net code is an individual string of five digits consisting of a GSM country code and a specific code for the network.

    This function is useful to prohibit unwanted "roaming", especially if the AS55X is working close to national borders when different networks are available.

    If the net code is unknown ask your GSM service provider or refer to www.gsmworld.com

    Default value: *not reported*

- Restart

    Referring to the global restart settings in "GSM settings / General" (see 5.9.1), you can here adjust every single channel.

    At this point the restart function can be activated ("Yes") or deactivated ("No") for this specific GSM channel. The parameters for the time interval and the operation mode have to be set globally in the window "GSM settings" / "General" (see 5.9.1).

    Default value: not reported

⚠️	Notice: A restart will never be triggered while a GSM connection or a call-setup is in progress.

⚠️	Notice: If the restart mode or the restart timer in the general window is switched off, the AS55X does not restart the timer if you choose "yes".
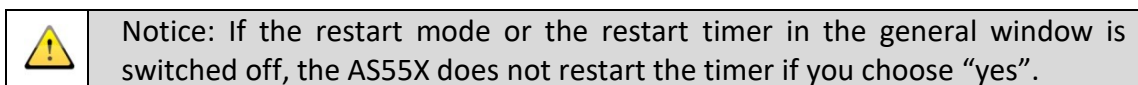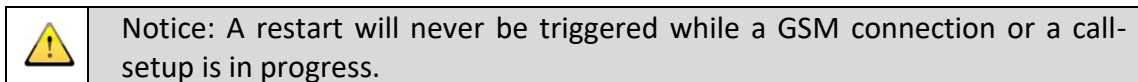
- Least signal quality

    To avoid poor audio quality or instable connections caused by a bad GSM signal level a threshold for a minimum GSM signal quality can be set. No call setup into the GSM network will be done via channels that do not reach this signal level.

    The indicator of a registered GSM channel with too small signal quality is a slowly flashing green LED.

    Adjustment range is   00% … 99%

    Default value: not reported

- Echo cancelling level

    In a plain ISDN PBX and additional use of ISDN telephone units no echo occurs. Because of the much higher speech comfort, echo cancelling should always be switched off in this case (operation mode "Duplex").

    In connections with analogue telephones in your PBX often echo phenomenon occur, which can be heard at the GSM phone. To avoid this, echo cancelling can be configured.

    There are three operation modes of the echo cancelling with different properties.
    - o 1st: Echo cancelling duplex (On low)
    - o 2nd: Echo cancelling simplex (On medium)
    - o 3rd: Echo cancelling simplex + duplex (On high)

    Echo cancelling duplex provides the smallest level of echo cancelling, but the best speech comfort. Echo cancelling simplex + duplex comes with the highest level of echo cancelling, but the worst speech comfort.

    Default value: Duplex

5.9.3   GSM groups

The AS55X has in the largest expansion stage up to 30 channels. The outgoing routing can become very complicated. Therefore multiple channels (e.g. the same provider) can be treated equally within groups. You can create up to 50 groups and rename them according to your requirements. For the selection of a channel to a group simply mark the corresponding box (an "X" appears). For the group name you only have to overwrite the default group name.

One channel can exist in multiple groups. If you want all channels in one group, use the button "Assign all GSM channels to this group". For every channel in its own group, use the second button "Assign GSM channel number to group number". Also you can clear one single group or alls groups by using the corresponding button.

5.10 GSM access

5.10.1 General

- Callback termination mode

  For more details about callback termination mode see 1.3.

  Here, the callback mode is selected. First you have to activate the callback function in "GSM access" / "by GSM channel" (see 5.10.2), secondly you have to choose between the two modes.

  If you choose "Reject call for callback after timeout" the incoming call will be rejected by the AS55X and a callback will be initiated immediately. The second options will be to "Accept call after timeout". Then you have to abort your call by yourself to activate a callback. Therefore you have to determine the phone call within the "call termination" time that is set in the field below. Otherwise, you can normally call an extension at your PBX and no callback will be initiated.

  Default value: *Reject call for callback after timeout*

- Call termination / acceptation timeout

  After this time the AS55X will reject the incoming call or within this time the incoming call has to be rejected manually by the call initiator.

  If you choose 0 seconds, the call will be rejected immediately.

  Adjustment range: 0 … 255 seconds

  Default value: *5 seconds*

- Maximum callback attempts

  If a callback to a mobile phone is unsuccessful, further attempts will follow until this maximum number. If it is not getting successful at all, the processing of a callback will be aborted.

  Adjustment range: 1 … 10

  Default value: *3*

- Callback delay

  After the call from a mobile phone is rejected for a callback or after the mobile phone has aborted his call in order to receive a callback, the AS55X will wait this time before starting the action.

  Adjustment range: 0 … 255 seconds

  Default value: *1 second*

- Callback repetition delay

    If a callback to a mobile phone has failed, the AS55X waits this time before starting the next attempt.

    Adjustment range: 0 … 255 seconds

    Default value: *15 seconds*

5.10.2 By GSM channel

There are two ways to handle the incoming calls from GSM. First you can make general settings for every GSM channel. If you now activate the option for "Access by number" you can specify your incoming routing just for any-known GSM number.

The global, general settings you can adjust here. For specify settings use the next window.

- Access from GSM permitted

    Here you can define whether GSM subscribers have access to the AS55X via this GSM channel or not. You can allow every user or user who submits the caller ID. By using the access list by GSM numbers, single or groups of phone numbers can be defined, which will be treated separately.

    Default value: *YES (AS551 and AS550)*

⚠️	Notice: Entries of the access list by GSM numbers have higher priority than this GSM channel dependent setting.

- Fix subscriber number

    Here you can configure a direct call to a PBX extension. All incoming calls on this channel will be directly forwarded to the fix subscriber number. This subscriber is called directly without any time delay and without any costs. If you use the subscriber delay, the call will be accepted by the AS55X and you have the options to call another PBX extension during this time delay by using DTMF suffix dialling. In this case the call is not free of charge.

    Please, also refer to point "Fix subscriber delay"!

    Default value: *No entry*

⚠️	Notice: By using the access list by GSM numbers, single or groups of phone numbers can be defined, which will be treated separately.

⚠️	Notice: Entries of the access list by GSM numbers have priority over the GSM channel dependent setting.

- Fix subscriber delay

  Normally if you use the fixed subscriber number, your call will be directly forwarded to the entered number every time. If you want to have the option to call another subscriber you can define a time delay until you will be forwarded to the fixed subscriber. During this time delay, you have the chance to call someone else by using the DTMF suffix dialling. If the AS55x detects no DTMF tones, the fixed subscriber is called after the chosen period.

  There are already traffic charges during suffix dialling.

  Please, also refer to point "Fix subscriber number"!

  Default value: *Off*

⚠️	Notice: By using the access list by GSM numbers, single or groups of phone numbers can be defined, which will be treated separately.
⚠️	Notice: Entries of the access list by GSM numbers have priority over the GSM channel dependent setting.

- Callback via CLIP

  If this feature is activated, the call from the GSM subscriber will be treated according to the callback settings in window "GSM access" / "General" (see 5.10.1). The calling mobile phone must have activated the display of its own number (CLIP). The callback code is not necessary for this feature!

  For more details about callback termination mode see 1.3.

  Default value: *No, deactivated*

⚠️	Notice: By using the access list by GSM numbers, single or groups of phone numbers can be defined, which will be treated separately.
⚠️	Notice: Entries of the access list by GSM numbers have priority over the GSM channel dependent setting..

- Callback via DTMF Code

  According to the "callback via clip" you have also the chance to create a callback by using CLIR. Therefore you have to activate this "callback via DTMF code" and the code must be linked to a GSM number in the list "by GSM number" (see 5.10.3). For entering, start your suffix dialling with the asterisk key " * " and then enter the callback code.

  The call to initiate callback via DTMF code is charged to the GSM subscriber!

  For more details about callback termination mode see 1.3.

  Default value: No, deactivated

- Callback via call-setup list

  If you have activated the callback for global or individual call-setups (see 5.8.2 and 5.8.3), the corresponding GSM channel have to activate here. The callback will work globally over all activated channels.

  For more details about callback mode see 1.1.

  Default value: No, deactivated

- Use access by number

  It is possible to set separately for every single GSM channel, if the access list by GSM number ought to be processed or not (see 5.10.3).

  By using of the access by GSM number, you can assign special GSM access data to single or groups of GSM subscribers. You are able to set callback, fix subscriber numbers and allow the access selectively. Furthermore you can assign callback codes to single numbers.

  The access list by GSM number will work as Whitelist, if an access mode is prevented in the access list by GSM channel and selectively allowed in the access list by number. The access list by number will work as blacklist, if an access mode is allowed in the access list by GSM channel and selectively prevented in the access list by number.

  An entry in the access list by GSM number always has higher priority than an entry in the access list by GSM channel.

  Default value: No, deactivated

- VPN Prefix

    If it is desire to work with a VPN in the GSM domain, a VPN prefix can be defined for every GSM channel. Please enter this prefix in international order beginning with "+".

    For additional information about the VPN prefix see 1.6.

    Default value: Not reported

5.10.3 By GSM number

You have the two opportunities to handle the incoming calls from GSM. First you can make general settings for every GSM channel. If you now activate the option for "Access by number" you specify your incoming routing just for well-known GSM number.

The global, general settings you can adjust in the window above. The specific settings you can set here.

In the list "Configure GSM access by number", you can enter up to 5000 values.

- (Beginning of) GSM number

    Here you can enter single GSM numbers or groups of numbers. The groups will affect all numbers with the same first digits (e.g. network prefix or VPN prefix). This groups and numbers can be entered continuously without special order. Longer numbers has a higher priority than shorter ones.

    The number has to be entered in international order, beginning with "+". A VPN number must be entered with the VPN prefix. For additional information about the VPN prefix, please see 1.6.

    All settings that are made to this individual number will overrun the global or general settings.

- Access permitted

    Here you can define whether GSM subscriber(s) has (have) access to the AS55X via GSM or not.

    Default value: Off

⚠	Notice: Entries of the access list by GSM numbers have priority over the GSM channel dependent setting.

- Fix subscriber number

  According to the channel access settings (see 5.10.2), you can also define a fix subscriber number for single numbers or groups of numbers.

  You can configure a direct call to a PBX extension. All incoming calls by this number or group of numbers will be directly forwarded to the fix subscriber number. This subscriber is called directly without any time delay and without any costs. If you use the subscriber delay, the call will be accepted by the AS55X and you have the option to call another PBX extension during this time delay by using DTMF suffix dialling. In this case the call is not free of charge.

  Please, also refer to point "Fix subscriber delay"!

  Default value: *No entry*

⚠️	Notice: Entries of the access list by GSM numbers have priority over the GSM channel dependent setting.

- Fix subscriber delay

  According to the channel access settings (see 5.10.2), you can also define special delays for single numbers or groups of numbers.

  Normally if you use the fixed subscriber number, your call will be directly forwarded to the inscribed number every time. If you want to have the possibility to call another subscriber you can define a time delay until you will be forwarded to the fixed subscriber. During this time delay, you have the chance to call someone else by using the DTMF suffix dialling. If the AS55x detects no DTMF tones, the fixed subscriber is called after the chosen period.

  There are already traffic charges during suffix dialling.

  Please, also refer to point "Fix subscriber number"!

  Default value: *Off*

⚠️	Notice: Entries of the access list by GSM numbers have priority over the GSM channel dependent setting.

- Callback via CLIP

    If this feature is activated, the call from the GSM subscriber will be treated according to the callback settings in window "GSM access" / "General". The calling mobile phone must have activated the display of its own number (CLIP). The callback code is not necessary for this feature!

    For more details about callback termination mode see 1.3.

    Default value: No, deactivated

⚠️	Notice: Entries of the access list by GSM numbers have priority over the GSM channel dependent setting.
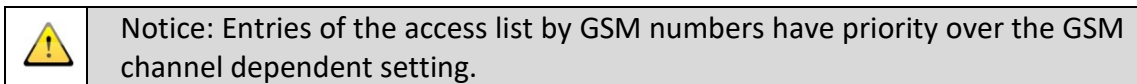
- DTMF code

    If you have activated the callback via DTMF code in "GSM access" / "By GSM number" (see 5.10.2) you can here define this DTMF code for single numbers. Consider that this code only make sense, if you have entered a single number. If you want to configure the same code for a group of numbers, you have to enter the numbers one by one. If you do not want this feature, leave this cell blank.

    Start your suffix dialling with the asterisk key " * " before entering the DTMF code.

    For more details about callback termination mode see 1.3.

    Default value: No entry

⚠️	Notice: Callback via DTMF code additionally has to be activated for every desired GSM channel!
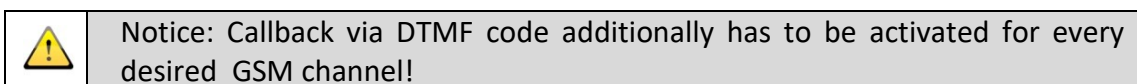
5.10.4 Internet

✋	Internet access is only available for WLL550 Pro.

Here you can configure necessary features for the internet access. This access is activated if the *"Go Online"* Box is marked or if APN details are inserted. The internet connection is a full time connection. It starts with system boot and you only can disconnect by switch off the WLL 550 Pro or by deleting the APN or unmarking the *"Go Online"* Box .

With firmware 3.20 the "Internet access data list" was introduced

Additionally to the hardcoded APN list there is way for a customized APN list. This list is useful in cases, the SIM could be changed during operation and the APN data of a provider could not be located in the hardcoded list. Furthermore there is a switch to allow or prevent data roaming.

Now there are four input fields where internet access data can be entered. If provider SimSwitch is activated, the entry according to the SIMs has the highest priority. The entry in wireless internet access screen has the next priority level followed by this new internet access data list. The hardcoded list got the lowest priority level. And if no matching data set could be found at all, internet access without APN will be tried.

For more details about internet mode see 0.
For LAN setup look here 5.2

⚠️	Notice: Check your tariff, because internet access is always up, if it is configured and the WLL 550 PRO is switched on.

- Access point name (APN)

  Here you have to enter your APN. The values you have to request by your provider.

- Username / Password

  As the authentication takes place via the SIM, in most cases username and password can be omitted. Otherwise you have to enter these values.

5.11 Extended features

5.11.1 PassThrough

For more details about PassThrough mode see 1.8.

✋	PassThrough is not available for the VOIP only gatways

5.11.1.1 General

If you want to use you your AS55X in PassThrough mode, you have to connect the AS55X between wired network and your PBX. Therefore the connection from wired network has to plug in ISDN main connector and the connection to PBX has to plug in-to ISDN aux.

⚠️	Notice: PassThrough is only possible if the AS55X works in ISDN mode.

- PassThrough operation mode

  o   Intercept calls from PBX

  In interception mode all dialled numbers to wired network will be checked by the AS55X. If there is a match with the "individual call-setup" list from "wired access" (see 5.8.3) and a designated GSM channel is available, the dialled number will be routed directly to wireless network.

  For special routings by calling party number or called party number you can configure complete numbers or number blocks. For more details please look in menu 5.8.3 ("wired access" / "individual call-setup").

⚠️	Notice: Entries in the individual call-setup list of wired access are mandatory for this function (see 5.8.3).

  o   MobileTwinning with calls from wired network

  Incoming calls from wired network can be routed to the wireless network additionally. Therefore you have to enter your wired and your mobile phone number in the "MobileTwinning list" (see 5.11.1.2). If there is a match in the twinning list, the AS55X will initiate a second call to mobile phone extension. Depending on where the call is accepted first, the other channel will be released.

  o   Interception and MobileTwinning

  In this mode, the two above options are running simultaneously.

- Code number for dialling from GSM to wired network / Dial code number to wired network

  To use your mobile phone as a kind of wireless extension of your PBX it is also possible to call from GSM to the wired network via AS55X, consequently also with the CLIP of your extension. Therefore you have to activate this code. The allocation of your CLIP will be made by the "MobileTwinning list" in the next window. All numbers which start with this code will not be routed to the PBX. For example all numbers with a leading zero will be routed to the wired network. Therefore it can be useful to mark the checkbox "Dial code number to wired network", otherwise the code will be cut off and the number will be transmitted without the leading zero. This is advisable if use a real code with up to four positions (e.g. 1234). Then you can cut off these digits. The code and the number has to be entered via DTMF.

  Default value: *Not reported / unchecked*

- Dialling tone GSM -> wired network

  This dialling tone will be generated by the AS55X, after the code number for dialling to the wired network has been entered. If the wired network generates a tone and if it shall be used, choose "Dialling tone from wired network".

  Default value: *Dialling tone from wired network*

- Idle tone GSM -> wired network

  This idle tone will be generated by the AS55X, when the called phone is ringing. If the wired network generates a tone and if it shall be used, choose "Idle tone from wired network".

  Default value: *Idle tone from wired network*

5.11.1.2 Mobile Twinning list

In the "MobileTwinning list" you can enter up to 5000 values.

- Wired number

  Here you have to enter your wired number to which the mobile number will be assigned. Mostly the wireless number and the wired number are in possession of the same user, so that the mobile phone and the wired phone will ring together.

  Normally the wired number has to be entered in international order, beginning with "+", because all dialled numbers will be converted to international numbers, if possible. If your provider transmits a wired number without prefix, you only have to enter your extension number.

- GSM number

  Here you have to enter your wireless number which is linked to your wired number.

  The number has to be entered in international order, beginning with " + ". A VPN number must be entered with the VPN prefix.

- Use this list for CLIP converting to wired network

  This entry is used to select a CLIP which is transmitted to wired network. If you call the GSM number of the AS55X and the dialling is directed to wired network, the AS55X checks if the GSM number is available in this list. If there is accordance to the GSM number the linked wired number is used as CLIP.

  Default value: *checked*

- Use this list for CLIP converting to PBX

  This entry is used to select a CLIP which is transmitted to your PBX. If you call the GSM number of the AS55X and the dialling is directed to your PBX, the AS55X checks if the GSM number is available in this list. If there is accordance to the GSM number the linked wired number is used as CLIP.

  Default value: *checked*

- CLIP converting for callback to wired network

  This entry is used to select a CLIP which is transmitted to wired network. If you use the callback function of the AS55X and you have a missed call from the wired network, the AS55X checks if your GSM number is available in this list. If there is accordance to the GSM number the linked wired number is used as CLIP.

  Default value: *checked*

5.11.2 SimSwitch

For more details about the AS55X SimSwitch see 1.9.

- Operation mode

    If you use an AS55X with SIM switch, you can choose for every channel up to six different operation modes. These six modes are three voice and three SMS modes.

    > ⚠️ Notice: If this operation mode is switched off, the channel will be treated as standard channel with one always activated SIM.

    > ⚠️ Notice: The modes for counting short messages are selectable only if Telnet is configured for the GSM channel.

    > ⚠️ Notice: All insert SIMs of one GSM channel must have the same (or no) PIN!

    o Only SIMs with inclusive minutes

    If you choose this configuration, all SIMs have the same tariff with the same amount of free inclusive minutes. The AS55X logs in the SIM cards one by one until all inclusive minutes are used. After all inclusive minutes are used up, no further outgoing connection setup to GSM can be made until a reset of the minute counters. If all amounts of free minutes are used for this channel the first SIM will be logged in again but without approving outgoing connections. Incoming calls are permitted if configured.

    > ⚠️ Notice: If all minutes are used up, the AS55X answers with ISDN cause / SIP response: "… the requested GSM channel is busy or not registered at the network or the signal quality is to bad"

    o SIM 1 with pay monthly tariff, others with inclusive minutes

    If you choose this configuration, all SIMs, without the main SIM, have the same tariff with the same amount of free inclusive minutes. The main SIM is with a monthly pay tariff. First the AS55X logs in the SIM cards with the inclusive minutes one by one until all inclusive minutes are used. After all inclusive minutes are used up, the main SIM will be logged in. This SIM will be activated for incoming and outgoing calls. After a reset of the minute counter, the SIM with the inclusive minutes will be logged in again.

    o Only SIMs with pay monthly tariff

    If you choose this configuration, all SIMs have the same tariff with pay monthly tariff. All SIMs will be logged in continuously by the entry "Switch to next SIM after … minutes respectively messages". This cycle is running endlessly.

    o Only SIMs with inclusive messages

If you choose this configuration, all SIMs have the same tariff with the same amount of free inclusive SMS. The AS55X logs in the SIM cards one by one until all inclusive messages are used. After all inclusive messages are used up, no further Telnet sessions can be made until a reset of the message counters. If all amounts are used for this channel the first SIM will be logged in again but without approving outgoing connections. Incoming calls are permitted if voice calls are configured.

- o SIM 1 with pay monthly tariff (SMS), others with inclusive messages

If you choose this configuration, all SIMs, without the main SIM, have the same tariff with the same amount of free inclusive messages. The main SIM is with a monthly pay tariff. First the AS55X logs in the SIM cards with the inclusive messages one by one until all inclusive messages are used. After all inclusive messages are used up, the main SIM will be logged in. This SIM will be activated for all further messages. After a reset of the message counter, the SIM with the inclusive messages will be logged in again.

- o Only SIMs with pay monthly tariff (SMS)

If you choose this configuration, all SIMs have the same tariff with pay monthly tariff. All SIMs will be logged in continuously by the entry "Switch to next SIM after … minutes respectively messages". This cycle is running endlessly.

- Total number of SIMs

Here you can enter the amount of SIM cards. These are calculated inclusive of the main SIM.

Allowed value: 1…5

> ⚠️ Notice: The presence of the main SIM is mandatory, additional SIMs must be filled up in rising order.
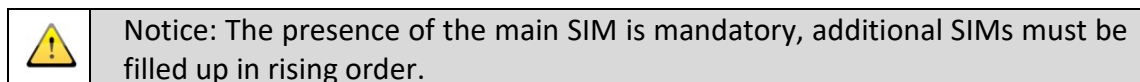
- No further connection after … minutes

During active calls, the airtime will be added up. After this limit has been reached, this SIM is blocked for outgoing connections. The call will not be aborted. But after the connection has finished, the next SIM will be activated. With reference to your average call duration, the limit should be a little under the inclusive minutes limit and below the "abort connection after … minutes" timer.

Allowed value: 1….65534

> ⚠️ Example: If you have a tariff with 1000 free minutes and your average call duration is about 10 Minutes, fill "985" in the "no further connection" timer and "999" in the "abort" timer. So the free minutes limit will not be crossed and no extra charges will be generated.
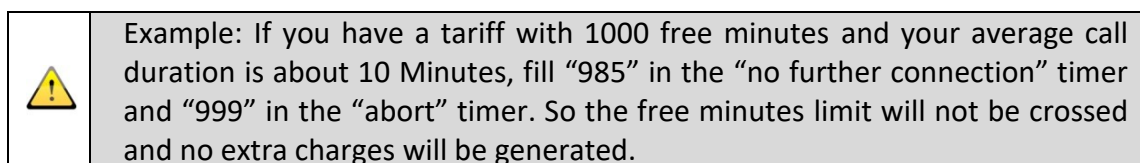
- Abort connection after … minutes respectively block SIM after … messages

Concerning to the timer above, you can here enter the limit, when the connection will be cut off. If you do not want to have extra tariff rates on your account, fill in here the inclusive limit of your SIMs. After the connection is aborted, the next SIM will be activated.

If you use this channel for SMS, you can fill in here your free limit. After this amount the SIM will be activated.

Allowed value: *1 … 65534*

⚠	Example: If you have a tariff with 1000 free minutes and your average call duration is about 10 Minutes, fill "985" in the "no further connection" timer and "999" in the "abort" timer. So the free minutes limit will not be crossed and no extra charges will be generated.

- Switch to next SIM after ... minutes respectively messages

    With this limit, it is possible that the cards will be permanently changed in a circle. The used minutes stay conserved until the minutes will be reset. In the operation mode "SIM 1 with pay monthly tariff, others with inclusive minutes / messages" the main SIM is excluded from this cycle.

    Allowed value: *1 … 65534*

- Step with call start (seconds)

    Some SIMs with inclusive minutes always counts an amount of seconds with the start of a call. With this setting, the size of the first step can be set. E.g. enter 60, is the first minute should always be counted full (60/1 clocking) or enter 1, if the counter should work linearly.

    Allowed value: *1...120*

- Further Steps (seconds)

    If the clocking of the SIM is in blocks even during the call, you can set the step width here.

    Allowed value: *1...120*

- Day of minute counter reset

    At this day of a month, all minute counters will be set to zero. Please pay attention to the settings of the AS55X internal clock! The minute counters will count the connected minutes only, if the connection setup direction is from the wired interface to GSM. If a month has fewer days than it is here, the reset will take place on the first day of the next month.

    Allowed value   *1...31*

⚠	Notice: If you want to have more information about your used minute or SMS amounts look at "Services" in the window "View SIM Switch status".

5.11.3 Telnet

For more details about Telnet see 1.10.

You can use up to two Telnet channels in the AS550 and up to six channels in the AS551 Master unit. The AS551 Slave units are not capable for Telnet channels.

If you use AS55X SMS server mode (see 5.11.4) you cannot use Telnet on this channel at the same time. All not useable channels are marked red.

- Telnet mode

    You can select between two different Telnet methods.

    First you have the "SMS and GPRS access" mode. In this mode data applications can run additionally and independently from normal voice operations.

    Secondly you can use the channel in "Full access" mode. In this mode you can use either Telnet or voice operations. The Telnet channel can only be opened if no voice connection is running or in progress. Voice connections cannot be started as long as the Telnet channel is open. When a Telnet session is terminated (e.g. the virtual COM port is closed), the GSM channel will fall back into voice mode and the indicator will show green again after a few seconds.

    Default value: *off*

⚠	Notice: If you use Telnet in full access mode, there will be no option for this channel to use incoming GSM remote maintenance.
⚠	Notice: If you use Telnet mode, the SimSwitch minute counter and the statistic will NOT count your connected minutes.

- Peer Telnet IP address

    If the Telnet access shall be locked to one dedicated client, fill in the IP address of this client here. Access from all other clients will be rejected.

    Different Telnet channels can be locked to different clients or to the same one.

    Default value: *Not reported*

⚠	Notice: This entry is strongly recommended to avoid illegal access to a GSM channel, especially if the AS55X can be reached from the internet.

- Local Telnet port

    Here you can enter the local Telnet port for every channel of the AS55X.

    Default value: *23 up to 28*

5.11.4 SMS server

For more details about SMS server see 1.11.

You can use every GSM channel for SMS server in the AS55X. It is also possible to use all AS551 Slave channel, so that up to 30 channels are available for SMS server.

If you use AS55X telnet mode (see 5.11.3) you cannot use SMS server on this channel additionally. All not useable channels are marked red.

- Telnet mode

    You can choice between two different SMS server methods.

    First you have the "SMS server multiple channels" mode. In this mode you can reach up to 30 channels via one telnet channel. Multiple channel mode is only possible on first Telnet channel. The other channel can be configured in single mode additionally. Voice traffic and SMS cannot be used parallel.

    Secondly you can use the channel in "SMS server single channel" mode. In this mode you can use SMS and voice at the same time. Up to all 30 channels can be used in this mode.

    Default value: *off*

- Peer Telnet IP address

    If the SMS server access shall be locked to one dedicated client, fill in the IP address of this client here. Access trials from all other clients will be rejected.

    Different SMS server channels can be locked to different clients or to the same one.

    Default value: *Not reported*

⚠	Notice: This entry is strongly recommended to avoid illegal access to a GSM channel, especially if the AS55X can be reached from the internet.

- Local Telnet port

    Here you can enter the local SMS server Telnet port for every channel of the AS55X.

    Default value: *23 up to 28*

⚠	Notice: For additional information ask your dealer for AS55X-SMS-API.

5.11.5 GSM-PBX

For more details about GSM-PBX see 1.12.

With this function you can achieve a direct link between a VOIP number and a GSM phone. You can register the AS55X directly with a VOIP provider, therefore the AS55X has to be SIP Client (see 5.7.1). You only need a broadband internet connection. If a call is established to your VOIP number, the associated phone will be called immediately or after a DTMF suffix dialling.

> ⚠️ Notice: You have to configure AS55X in SIP client mode for GSM-PBX. GSM PBX is not available in ISDN mode.

5.11.5.1 Registration

There is no difference between your data here or in the menu "SIP Access" / "Names and passwords" (see 5.7.3). If you have different usernames for each SIP number use the link list.

- SIP Provider hostname, Registrar name

    Here you have to enter your provider´s domain name or name of registrar. If you enter the local hostname and the remote hostname in "SIP access" / "Names and passwords", it will be overwritten by this value. If you had entered there the local hostname before, it will be filled in here.

    If this field is empty, the IP address of the AS55X is insert instead and the entry "Registrar IP address" or "Registrar hostname" in "SIP access" is mandatory.

    Default value: *not reported*

- Local username for SIP-DID registration

    If SIP-DID registration is required, fill in your account data which you have achieved from your provider. Often username and authentication ID for registration are identical. In this case, the authenticate ID field below, can be left empty.

    This entry is the same as the local username in the menu "SIP Access" / "Names and passwords".

    This entry is case sensitive.

    Default value: *not reported*

- Authenticate ID for SIP-DID registration

  If your local username and your authentication ID are not the same you have to enter it here. Often authentication ID and username for registration are identical. In this case, this field can be left empty.

  This entry is the same as the local username in the menu "SIP Access" / "Names and passwords".

  This entry is case sensitive.

  Default value: *not reported*

- Authenticate password for SIP-DID registration

  Here you have to enter your password of your account data.

  This entry is the same as the local username in the menu "SIP Access" / "Names and passwords".

  This entry is case sensitive.

  Default value: *not reported*

- Contact username for SIP-DID registration

  The standard contact username for SIP-DID registration is "*". If your provider needs another one, please fill in here. If this field is left empty, "*" will be used.

  Leave this entry empty in case of single user registrations.

  Default value: *not reported*

5.11.5.2 Link list

In the Link list you can enter up to 100 different SIP numbers. For each number you can define if it is linked directly to a GSM number, if you can use DTMF suffix dialling or if the number is not available for incoming calls. If you use a direct link to a GSM number, the link is double sided. On the one hand your mobile Phone is directly connected to the SIP number. On the other hand you can call the AS55X via GSM and dial a number on fixed network via DTMF suffix dialling (or via DTMF speed dialling). Therefore the linked SIP number will be used for the outgoing call (especially the outgoing CLIP). All GSM users, who are not listed, will use the default outgoing identity which you have to select.

- SIP number

    Here you have to enter your SIP number. Enter this number in order your SIP provider allocated it to your account.

    This number may consist of numbers and letters.

    Default value: *not reported*

- Authenticate ID / Password

    If you have to register your SIP accounts with different logins (authentication and password) you have to fill in here. If all SIP accounts use the same notification, you have to enter this global ID in the window "SIP access" / "Names and passwords".

    Default value: *not reported / not reported*

- Default outgoing identity

    You can define one SIP number as default outgoing identity. If a call from GSM to SIP cannot be assigned via GSM CLIP to one entry of this list, this default identity is used for a call setup to the SIP provider.

- Link

    With this selection, you decide how a call from SIP site has to be handled. If you set "None", the call will be rejected. With "DTMF suffix dialling", the caller gets a tone or announcement as set in configuration form "General" / "Tones and announcements" (see 5.1.1) and he can dial a DTMF number string. This can be a DTMF number of the DTMF speed dialling list or a complete GSM number.

    Default value: *not reported*

- GSM number

  Here you have to enter the linked GSM number. This number should be entered in international order, beginning with "+". A VPN number must be entered with the VPN prefix.

  For the call setup direction to GSM, this call is directly forwarded to this GSM number if "Link to GSM number" is set. For the call setup direction to SIP, this number used to select the outgoing identity.

  The GSM number and its linked SIP number is also used by the DialCommand feature (see 5.11.4)

5.11.5.3 DTMF speed dialling list from SIP

For quick and easy dialling from SIP you can save up to 100 phone numbers in your AS55X which you can dial by a DTMF shortcut. For these abbreviations you can use all twelve keys on your phone, including hash "#" and the asterisk "*". The length of this DTMF string is limited to 21 characters. By default the first match of your shortcut is selected. A precise adjustment of the selection takes place in the menu "DTMF dialling options".

- DTMF number

  This is the abbreviated dial string, received by DTMF from the wired calling entity, which has to be converted into a complete GSM number.

  The digits "0" … "9" as well as "*" and "#" are allowed

⚠	Notice: It´s additional possible to dial a mobile phone number in this mode. You don´t have to configure this additionally. If there is no match in the DTMF list, the entered number will be dialled. Note also the menu "DTMF dialling options".

- GSM number

  The number should be entered in international order, beginning with "+". A VPN number must be entered with the VPN prefix.

5.11.5.4 DTMF speed dialling list from GSM

For quick and easy dialling from GSM you can save up to 100 numbers in your AS55X which you can dial by a DTMF shortcut. For these abbreviations you can use all twelve keys on your phone, including hash "#" and "*". The length of this DTMF string is limited to 21 characters. By default the first match of your shortcut is selected. A precise adjustment of the selection takes place in the menu "DTMF dialling options".

> ⚠ Notice: Calls from GSM have to be permitted in the "GSM access" / "by GSM channel" (see 5.10.2) menu for every GSM channel.

- DTMF number

  This is the abbreviated dial string, received by DTMF from the wired calling entity, which has to be converted into a complete SIP number.

  The digits "0" ... "9" as well as "*" and "#" are allowed

> ⚠ Notice: It´s additional possible to dial a SIP number in this mode. You don´t have to configure this additionally. If there is no match in the DTMF list, the entered number will be dialled. Note also the menu "DTMF dialling options".

- GSM number

  The number should be entered in international order, beginning with "+". A VPN number must be entered with the VPN prefix.

5.11.5.5 DTMF dialling options

For incoming calls from GSM and from VOIP you can activate a direct call or use DTMF suffix dialling. For incoming VOIP calls you have to activate the "DTMF suffix dialling" in the "link list", for calls from GSM you have to permit access from GSM in the menu "GSM access" / "By GSM channel" (see 5.10.2) or "By GSM number" (see 5.10.3). For both directions you can use the DTMF speed dialling with the linked numbers in the menus above. But if speed dialling is activated you can also dial numbers outside this list. By default all numbers you have dialled and which do not match to this list will be forwarded. To accelerate this, you can optimize the detection with the following setting.

- Maximum amount of digits for dialling from SIP

  A dialled number will be detected as complete after this amount of digits from SIP. Then the call setup will start immediately.

  Default value: *11*

- Detect dialling from SIP after... or with DTMF digit ...

  A dialled number from SIP will be detected as complete after this duration or after this DTMF digit. For termination with a DTMF digit you can use " # ", " * " and the digits " 0...9 ". After one of these two events, the call setup will start immediately.

  Default value: *4 seconds / " # "*

- Check speed dialling list with every digit from SIP

  Normally with every received digit, the dialled number will be searched in the regarding speed dialling list. But if it should be possible to dial long numbers that have the same beginning as whole short numbers, this box must be unchecked. In this case, the speed dialling list will be checked after the dialled number is detected as complete.

  Default value: *checked*

- Allow complete number dialling from SIP

  If dialling of complete GSM numbers is allowed, protection against misuse is strongly recommended. This can be done by restricting the access to GSM to a couple of dedicated users. Therefore use the menu "Sip access" / "Authorized user list" (see 5.7.4).

  Default value: *checked*

- Default destination if invalid or no dialling from SIP

  Fill in a destination number to be called, if no DTMF dialling is made by the calling subscriber within 15 seconds.

  Default value: *not reported*

- Maximum amount of digits for dialling from GSM

  A dialled number will be detected as complete after this amount of digits from GSM. Then the call setup will start immediately.

  Default value: *11*

- Detect dialling from GSM complete after ... or with DTMF digit ...

  A dialled number from GSM will be detected as complete after this duration or after this DTMF digit. For termination with a DTMF digit you can use " # ", " * " and the digits " 0...9 ". After one of these two events, the call setup will start immediately.

  Default value: *4 seconds / " # "*

- Check speed dialling list with every digit from GSM

  Normally with every received digit, the dialled number will be searched in the regarding speed dialling list. But if it should be possible to dial long numbers that have the same beginning as whole short numbers, this box must be unchecked. In this case, the speed dialling list will be checked after the dialled number is detected as complete.

  Default value: *checked*

- Allow complete number dialling from GSM

  If dialling of complete VOIP numbers is allowed, protection against misuse is strongly recommended. This can be done by restricting the access to VOIP to a couple of dedicated users. Therefore use the menu "GSM access" / "by GSM channel" (see 5.10.2) and "by GSM number" (see 5.10.3)

  Default value: *checked*

- Default destination, if invalid or no dialling from GSM

  Fill in a destination number to be called, if no DTMF dialling is made by the calling participant within 15 seconds.

  Default value: *not reported*

⚠	Notice: Settings in menu "By GSM channel" or in menu "By GSM number" will overrun this setting.

5.11.6 SNMP

For more details about SNMP see 1.13.

You can configure SNMP in tow different modes and for two different SNMP managers.

- SNMP Port

    This UDP port is used by the SNMP agent of the AS55X for receiving and sending of SNMP messages.

    Default value: 161

- Write value for statistic reset

    If the reset of the AS55X statistic should be permitted via SNMP, there must be defined a write value. If this is written to the SNMP variable "statisticReser", all current statistics values will be deleted and all statistic counters will restart.

    Default value: *not reported*

- Operation mode

    Please define if you want to permit SNMP read operations from and/or write operations to the AS55X. Additionally you can configure the AS55X to sent SNMP traps to the manager.

    Default value: *Off*

- IP address

    This IP address is used to identify SNMP manager for read and/or write operations and has to be entered. This IP address is also used to send traps.

    Default value: *not reported*

- Trap port

    This UDP port is used by the SNMP agent of the AS55X so address SNMP messages.

    Default value: *not reported*

- Read community

    Here you can define a community string for read community (e.g. "public" or "private"). It is used to identify the SNMP manager.

    Default value: *not reported*

- Write community

  Here you can define a community string for write community (e.g. "public" or "private"). It is used to identify the SNMP manager.

  Default value: *not reported*

With firmware 3.20 we introduce some new features for SNMP.

- The mobile network type, if GSM, EDGE, UMTS or LTE, can now be requested via SNMP.

- By default the GET-NEXT requests within the table firstly increment the table element and when the whole table is processed, the index will be incremented.
  This order can be changed by setting the unnamed bit 45.

5.11.7 PassThrough backup mode

For more details about the PassThrough backup mode see 1.14

If the connection to the fixed line network is lost, all calls can be routed via GSM. For this feature, an additional license is necessary.

Please select here, if calls should be routed via GSM network if the connection to fixed line network is lost completely or if the network cannot route the calls due to congestion or resource problems.

Please keep in mind that the call-setup time will be longer in this case because the AS55X first tries to forward the call to the fixed network. It has to check via timer if all dialled digits arrived and finally starts the call via GSM.

Default value: *not reported*

5.11.8   ProviderSimSwitch

✋	Provider SimSwitch is only available for AS55x and WLL 550 Pro with SimSwitch hardware.

For more details about the ProviderSimSwitch see 1.15

To configure the "ProviderSimSwitch" activate the SIM slot of SIM 1 to 5 by setting it to "Yes". Please begin with SIM 1 and do not skip any SIM slots such as using SIM 1, SIM 2 and SIM 4. Each GSM channel can manage different numbers of SIM cards. Entering the GSM netcode is optional to avoid international roaming.

This example is a setup for WLL550 Pro with two SIM cards for each GSM channel. APN information is only necessary for GSM channel 1. The GSM channel 2 does not support LTE or UMTS with HSPA+ or HSDPA.

5.11.9   Fax extension

✋	The Fax extension is only available for AS55x and WLL 550 Pro with SIP capabilities.

For more details about the Fax extension please contact the support.

To configure the "Fax extension" a working SIP setup is mandatory. The "Fax extension" is an additional SIP extension number. It is recommended to assign the service to a fix GSM-channel. Within the WLL 550 Pro this is always the channel 2.

Default value: *not reported*

5.12 Special settings

⚠	Notice: To view this menu you have to start the expert mode. Therefore open the tab "Preferences in the menu bar.

⚠	You can make special settings with this tool, which can be necessary in some extraordinary cases. In normal cases this special configuration should not be used! Unwanted behaviour of the AS55X will be consequence, if you make changes without exactly knowing the function of the settings. Please contact your distributor before using this tool

5.12.1 Timers

- Call from ISDN reaction delay

  In very few cases PBXs have problems if the reaction to a SETUP is too fast. A delay can be defined at this point.

  Default value: *0.10 sec.*

- Waiting time for first digit from ISDN

  The AS55X waits this time for the first digit of ISDN suffix dialling. After this timer has expired, the call-setup attempt from ISDN domain will be aborted.

  Default value: *15 sec.*

- Waiting time for reconnect from ISDN

  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.

  Default value: *120 sec.*

- Time limit of callback attempt to ISDN

  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.

  Default value: *60 sec.*

- Call from GSM reaction delay

  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.

  Default value: *1.00 sec.*

- Waiting time for CLIP from GSM

  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.

  Default value: *2 sec.*

- Waiting time for first DTMF digit from GSM

  The AS55X waits this time for the first digit of DTMF suffix dialling 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.

Default value: *15 sec.*

- Waiting time for next DTMF digit from GSM

  The AS55X waits this time for the next digit of DTMF suffix dialling from the mobile phone. After this timer has expired, the call-setup attempt from GSM domain will be aborted.

  Default value: *15 sec.*

- Time limit of call to fix subscriber

  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 dialling.

  Notice: If the caller is enabled for DTMF dialling after an undelayed fix subscriber call has been made, there will be traffic charges to the mobile phone during suffix dialling.

  Default value: *60 sec.*

- Waiting time for connect from GSM

  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.

  Default value: *120 sec.*

- Blocking time of GSM channel after call termination

  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.

  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.

  Default value: *1.0 sec.*

- Delay before start of GME active Task

  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.

  Default value: *0.50 sec.*

- Get GSM cell information repetition time in idle mode

  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.

  The distances of reading the cell information in idle mode and in active mode (call present) can be set separately.

  The information can be displayed via the trace functions.

  Default value: *0 sec.*

- Get GSM cell information repetition time in active mode

  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.

  The distances of reading the cell information in idle mode and in active mode (call present) can be set separately.

  The information can be displayed via the trace functions.

  Default value: *0 sec.*

- SIP OPTIONS request repetition period

  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.

  With this setting, the distance of two OPTIONS requests can be specified.

  Default value: *10.00 sec.*

- SIP registration repetition

  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.

  Default value: *300 sec.*

5.12.2 Unnamed bits

- Unnamed bit 3, reset GSM channel

  If this bit is set, the GSM channel will be restarted when an outgoing call is aborted within one second or with error code 'call barred'.

- Unnamed bit 5, fixed link of ISDN B-channel to GSM channel

  If this bit is set, in both directions from ISDN to GSM and from GSM to ISDN the ISDN B-channel are directly linked to the correspondent GSM channel. With this option, the GSM channel routing can be made by the PBX.

  In AS551 BRI 4V, the left BRI interface pair is linked to GSM channel 1 (B-channel 1) and 2 (B-channel 2) and the right interface pair is linked to GSM channel 3 (B-channel 1) and 4 (B-channel 2).

- Unnamed bit 7, do not process national emergency numbers

  If set, national emergency numbers are not extra processed. This bit has no effect to the GSM emergency number 112.

  Default value: *Set*

- Unnamed bit 9, trace GSM engine start-up sequence

  If set, the start-up sequence of the GSM engines will be traced. This bit can be used to decrease trace load if modules are missing SIMs.

  Default value: *Set*

- Unnamed bit 10, use GSM network nationality

  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.

  Default value: *Not set*

- Unnamed bit 11, allow callback to GSM before callback via call setup list

  If both callback via call setup list and callback to the GSM subscriber apply to one call from GSM, the callback to the GSM subscriber will be executed first if this bit is set. If it is not set, the callback GSM will be suppressed in this case.

  Default value: *Not set*

- Unnamed bit 14: Load audio parameters unconditional

  If set, the audio parameters for the GSM module will be loaded unconditional with every module start-up.

  Default value: *Not set*

- Unnamed bit 15: Never load audio parameters

  If set, the audio parameters for the GSM module will never be loaded.

  Default value: *Not set*

- Unnamed bit 16, GME code suppression

  If set, the DTMF tones for the GME code in direction to wired interface will be suppressed.

  Default value: *Set*

- Unnamed bit 17, write information element channel identification unconditional

  If set, this ISDN information element will be included in every call setup message. Otherwise, it will be included according to ETSI rules.

  Default value: *Not set*

- Unnamed bit 26, use address placeholder in via header

  If set, the actual IP address and port number are not included in SIP via header. They are replaced by placeholders. This bit setting might be necessary, if a NAT router is placed between AS55X and PBX (or SIP provider).

  Default value: *Not set*

- Unnamed bit 28, allow active SIP call without audio

  If set, an available audio channel is not necessary for successfully setting up a SIP call. This setting is helpful, if the PBX defines the audio channel not till the call is active.

  Default value: *Not set*

- Unnamed bit 29, redirect RTP stream to received address / port

  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).

  Default value: *Set*

- Unnamed bit 31, SimSwitch count minutes instead of seconds

  Helpful for SimSwitch demonstration or evaluation. If set, the second counter are running 60 times faster.

  Default: Not set

- Unnamed bit 31, SimSwitch count minutes instead of seconds

  Helpful for SimSwitch demonstration or evaluation. If set, the second counter are running 60 times faster.

  Default value: *Not set*

- Unnamed bit 32, keep remote logical address header for moved

  This bit must be set for SIP call forwarding or redirection with "moved".

  Default value: *Not set*

- Unnamed bit 34, detect "+CMUX:" as sent short message

  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.

  Default value: *Not set*

- Unnamed bit 35, suppress DHCP request as connection check

  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.

  Default value: *Not set*

- Unnamed bit 36, deactivate local host queue

  Test setting: LAN packet, sent to the own IP address will not be handled internally and they will be sent via the LAN.

  Default value: *Not set*

- Unnamed bit 40, accept any OPTIONS response

  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.

  Default value: *Not set*

- Unnamed bit 46, do not allow SIP IP and port redirection

  There is 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.

  Default value: *Not set*

- Unnamed bit 47, use local username for SIP registration

  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.

  Default value: *Not set*

- Unnamed bit 48, get public IP address from register response

- Unnamed bit 49, get public port from register response

  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 this information from the register response message can be controlled.

  Default value bit 48:   *Not set*
  Default value bit 49:   *Not set*

- Unnamed bit 50, abort SIP connection with TCP socket close

  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.

  Default value: *Not set*

- Unnamed bit 52, allow registrar IP as proxy identity

  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.

  Default value: *Not set*

- *Unnamed bit 55,* FAX speed restriction

  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.

  Default value: *Not set*

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- Unnamed bit 56, no FAX function restriction

    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.

    Default value: Not set

- Unnamed bit 59, do not store local parameters

    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.

    Default value: Not set

- Unnamed bit 60, do not store remote parameters

    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.

    Default value: Not set

- Unnamed bit 65, add a leading '0' zero to called party number

    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.

    Default value: Not set

- Unnamed bit 66, do not park GME extensions

    In GME mode, normally extensions are parked if a second one is dialed. If this bit is set, the extension will not be parked and voice channel will be kept.

- Unnamed bit 67, use any channel if selected group busy

    If for an outgoing call all channels of the designated group are busy, one of the   other ready channels will be used. Hence, the selection of the group is not   exclusively,   it   is preferred.

- Unnamed bit 68, suppress SIP PROGRESS

    Some PBXs cannot interpret SIP PROGRESS. With this bit, it will be suppressed.

- Unnamed bit 69, suppress SDP in SIP RINGING

  Some PBXs do not want to find SDP in SIP RINGING. With this bit, it will be suppressed.

- Unnamed bit 70, allow SIP remote logical address header redefinition

  In single cases, PBXs change the local tag in the To-Header without permission. With setting of this bit, the AS55X will tolerate it.

- Unnamed bit 74, suppress CLIP with FAX calls

  Normally the GSM number for FAX calls differs from that one for voice calls. But for outgoing FAX calls, the voice number is displayed to the recipient. Hence, a FAX back to the displayed number will fail. With this bit, no number is displayed, so the only information for a FAX back is the identification in the head line of the FAX.

- Unnamed bit 75, convert DTMF hash codes

  Dialing strings beginning with a '#' are used for programming or switching purposes. If dialing of star-hash codes to the network is required, this bit can be used. If it is set, the DTMF digit sequence '1#012345' will be converted to '#012345' and '2#012345' will be converted to '##012345'.

- Unnamed bit 76, ignore network selection result

  See unnamed byte 126.

5.12.3 Unnamed bytes

- Unnamed byte 12, traffic direction

  These bytes specify the allowed call setup directions:
  1: Calls from PBX to GSM are allowed
  2: Calls from GSM to PBX are allowed
  3: Calls in both directions are allowed

  Default value: *3*

- Unnamed byte 19, overrun type of callback number

  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.
  0: Unknown type
  1: Subscriber type
  2: National prefix and digits
  3: National type and digits
  4: International prefix, country code and digits
  5: International type, country code and digits

  Default value: *255 (decimal), no overrun*

- Unnamed byte 29, DTMF processing

  This byte holds the configured DTMF processing modes.
  1: Detect restart sequence (*0) during call setup from GSM to PBX
  2: Detect restart sequence (*0) from GSM during active call
  4: Convert SIP telephone event to DTMF to GSM
  8: Convert DTMF from GSM to telephone event to SIP
  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.

  Default value: *3*

- Unnamed byte 56, call waiting off trials

  This byte specifies the trials of switching call waiting off with the start-up procedure of the module. If the GSM network does not accept this command, the start-up of the module gets faster if this value is set to 0.

  Default value: *1*

- Unnamed byte 59, GME timeout passive line

  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.

  Default value: *120 (decimal)*

- Unnamed byte 63, type of CLIP to ISDN for AUX interface

  Same as with "Type of CLIP to ISDN", settable with general ISDN access but not for the main interface.
  0: Unknown type
  1: Subscriber type
  2: National prefix and digits
  3: National type and digits
  4: International prefix, country code and digits
  5: International type, country code and digits

  Default value: *2*

- Unnamed byte 66, SIP server registration mode

  1: SIP clients have to register explicit with REGISTER request (normal mode)
  2: SIP clients can also register implicit with an INVITE request

  Default value: *1*

- Unnamed byte 69, IP trace mode

  1: Trace received frame , 2: Trace sent frame , 4: Trace received fragments

  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".

  The trace load can be reduced with this byte, e.g. if a deeper trace memory for GSM is desired.

  Default value: *0*

- Unnamed byte 70, SIP trace mode

  1: Trace received frame if parse error
  2: Trace unexpected received messages
  4: Trace received messages unconditional
  8: Trace transmitted frames
  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 desired.

  Default value: *255 (decimal), all modes active*

- Unnamed byte 75, overrun LAN chip settings (Since firmware 3.00 via menu)

  Hex 80: 10 Mbps, half duplex
  Hex 81: Negotiation 10/100Mbps, half/full duplex
  Hex 82: 100 Mbps, half duplex
  Hex 83: Negotiation 10/100Mbps, half/full duplex
  Hex 84: 10 Mbps, full duplex
  Hex 85: Negotiation 10/100Mbps, half/full duplex
  Hex 86: 100 Mbps, full duplex
  Hex 87: Negotiation 10/100Mbps, half/full duplex

  By default auto negotiation is used to avoid a traffic bottle neck in the switch. This can occur if the switch port, the AS55X belongs to, works with 10 Mbps and all others with 100 Mbps or more.

  If there are problems with lost packets, this value can be changed to 10 Mbps. In worst case with 30 active calls via AS551 there is only a maximum LAN load of about 3 Mbps.

  Default value: *Hex 81*

- Unnamed byte 77, SIP REFER acknowledge mode

  0: Acknowledge with OK
  1: Acknowledge with NOTIFY

  Default value: *0*

- Unnamed byte 78, offered telephone event payload type

  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).

  Default value: *0 (same as 123)*

- Unnamed byte 82, SIP phone context processing

  The SIP phone context is an additional information element of the SIP FROM or TO headers in order to include number prefixes.
  With the value 1 of this byte, phone context is used, if the phone number is not in national or international order.
  With the value 2, phone context will be used unconditional and a national or international prefix of the phone number will be removed.

  Default value: *0, phone context processing off*

- Unnamed byte 87,

    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 forwarding, it might be possible to delay this rejection. A delay can be entered in seconds.

    Default value: *0, immediately*

- Unnamed byte 90, TCP message distance

    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.

    Default value: *0*

- Unnamed byte 91, minimum DTMF tone length from GSM

    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.

    Default value: *0, same as 12 (48 ms)*

- Unnamed byte 103, Telnet option raw TCP mode

    If this byte is set to 1, no Telnet control codes are used. This is mostly called 'raw TCP mode'.

    Default value: *0*

- Unnamed byte 107, read own voice number trials

    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.

    Default value: *5*

- Unnamed byte 109, pause between FAX call-setup trials

    If a FAX call-setup failed, after this time in seconds, the WLL is trying again.

    Default value: *0, same as 60*

- Unnamed byte 110, pause between FAX call-setup trials after connection

    If a FAX connection failed, after this time in seconds, the WLL is trying again.

    Default value: *0, same as 10*

- Unnamed byte 113, maximum call trials to local FAX

    The maximal number of call trials to the local FAX device, that is connected to the analogue interface of the WLL.

    Default value: *0, same as 3*

- Unnamed byte 114, maximum call trials to remote FAX

  The maximal number of call trials to the remote FAX device, that is connected via GSM to the WLL.

  Default value: *0, same as 3*

- Unnamed byte 115, minimum DTMF tone length from wired side

  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.

  Default value: *0, same as 12 (48ms)*

- Unnamed byte 118, remove leading '0' zero from calling number

  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.
  If this byte is 1, a leading zero will be removed from an unknown number.
  If this byte is 2, a leading zero will be removed from an ISDN national number.
  If this byte is 4, a leading zero will be removed from an international number.
  If this byte is 7, a leading zero will be removed from all number formats.
  A value of 128 deactivates this function.

  Default value: *0, previous behaviour*

- Unnamed byte 119, ignore callback digits if PBX internal number

  If the PBX sends the CLIP of the extension with leading digits (e.g. local area code) and if it expects the called number for callback without this leading digits, the number of digits to be removed can be set here.

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- Unnamed byte 121, force UMTS or LTE modules to a dedicated network type

    Normally these modules preferably registers to the network with highest bandwidth. If byte 121 is set to a value other than zero, the channel rate will be fixed: 1 for GSM, 2 for UMTS and 7 for LTE. Other values depend on the module types. This setting is globally for all channels.

    Default value: 0

- Unnamed byte 123, hold line on wired side

    It can be necessary to hold the line on wired side when the call is terminated   form the GSM side. With this byte the termination of the call on the wired side can   be   delayed for this number   of seconds.

- Unnamed byte 124, registration timeout

    By default the timeout for registration of the GSM-module to the network is 120 seconds. This byte can be used for another value.

- Unnamed byte 126, network selection trials

    By default, this value is 8. It can be increased, if the registration of the TC63i GSM module to the network fails for a very long time, e.g. in the case of roaming. If the unnamed bit 76 is set, error results will be ignored

- Unnamed byte 127, SIP instant messaging response type

    0: The SIP MESSAGE request will be answered immediately with the final response 202 (Accepted), if the message is accepted for sending (a GSM-channel is ready, number and message are valid).
    1: The SIP MESSAGE request will be answered immediately with the provisional response 100 (Trying) and then with the final response 200 (OK) if the sending is successful.
    2: The SIP MESSAGE request will be answered with the final response 200 (OK) if the sending is successful.

5.12.4 Unnamed words

- Unnamed word 15, SIP local RTP port base

  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.

  Default value: *30000*

- Unnamed Word 16 to 18, Ethernet MAC address overrun

  Default value: *0, factory setting*

- Unnamed word 19, SIP REGISTER repetition period

  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 value: *300 (5 minutes)*

- Unnamed word 20, SIP OPTIONS repetition period

  The AS55X periodically sends an OPTIONs message, the period length can be changed here. In SIP peer mode, the sending begins immediately after start-up, 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 value: *1000 (10 seconds)*

- Unnamed word 21, service port

  This is the listen port for service access via LAN. The default port is 29999 (TCP).

  Default value: *0 (same as 29999)*

- Unnamed word 22, jitter buffer initial delay

  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 value: *0 (same as 50 ms)*

- Unnamed word 24, repetition period of NAT keep alive messages

    This value is insinuating in 10 ms steps.

    Default value: *0 (same as 1000, 10 seconds)*

- Unnamed word 26, outgoing local TCP port range begin
- Unnamed word 27, outgoing local TCP port range end

    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.

    Default value word 26:   *10000*
    Default value word 27:   *10999*

5.12.5 Unnamed long words

- Unnamed long word 10, SIP CLIP source header list.

    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.
    Digit value 1: From header
    Digit value 2: P-Asserted-Identity header
    Digit value 3: P-Preferred-Identity header
    Digit value 4: Contact header
    Digit value 5: Remote-Party-ID
    Other values are not defined.
    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.

    Default value: *00000123*

5.13 Test licences

⚠️	Notice: To view this menu you have to start the expert mode. Therefore open the tab "Preferences in the menu bar.

The manufacturer grants a test licence for maximum device features for a limited time of four weeks (28 days). This duration can be divided into an infinite number of parts. After this test licence has been started, it can be halted in order to start it again later on. With the test licence, the AS55X can be configured like with regular licenses. After the license duration has expired, the device stops normal processing and only maintenance is possible. All SIP Clients and SIM cards will be stay logged in but all calls will be declined. Calls will not be aborted. There will be no LED notification. The status of the test licence you can check up in menu 7.16. Now the configuration has to be up- and downloaded in order to make the AS55X restart without test licence and without the configuration part of the extended features of the test license. If you want to obtain all configuration data, including the test features, login and save your configuration data first or fill in an upgrade code. All data are kept until the first configuration download to AS55X.

6 Installation

6.1 Set SIM PIN

If your inserted SIM Cards need a PIN to log in the wireless network, you have to enter it here. You can enter a different SIM-PIN for every channel. If you use a SimSwitch, it is mandatory to use the same (or no) PIN for all SIMs in one channel.

To save the PIN in AS55X, enter it, select the channel and press "Apply". The PIN will be saved in the device and NOT in the configuration data. If you use this configuration data in a second device or backup it, the PINs will not be saved. You have to enter the PINs again.

If you enter a wrong PIN, the LED will flash five times red. If you do this twice, the LED will flash six times. The third and last attempt is not possible in AS55X. Therefore you have to put your SIM in a plain mobile phone. If you enter no PIN but the SIM requires one, the LED will flash four times. These SIM messages you can also check in Services menu (see 7.2).

6.2 Set password local access

Here you can assure your AS55X against unauthorized access with the AS55X-Service-Software via USB. By default no password is set.

To set a password for first configuration, leave the current password blank and type in the new password inclusive the verification. To delete a current password, type in your old password and leave the other two fields blank.

The password is saved in the AS55X and NOT in the configuration data.

⚠️	Notice: If you lose your password, it can be cleared by the manufacturer only.

6.3 Set password remote / LAN access

Here you can protect your AS55X against unauthorized access via LAN and remote access with the AS55X-Service-Software via GSM. Defining the username and password is recommended. To avoid any unwanted LAN or GSM access, switch off the remote access switch at the front (AS551) or back (AS550) of your gateway. By default no username and no password is set.

To set a username/password for first time, leave the current password blank and only type in the new password inclusive the verification. To delete a current password, type in your old password and leave the other two fields blank. For changing the username the method is the same.

The username and password are saved in the AS55X and NOT in the configuration data.

⚠️	Notice: If you lost your username and password, it can be cleared by the manufacturer only.

6.4 Set name of device

For better determination and for ease of operation, you can assign a name for each AS55X device. This will appear in the search for a device in the LAN or at the services menu in the device data

The name of the device is not part of the configuration and will not be saved in configuration data / file.

6.5 Download firmware

Here you can update your AS55X. The update will be transferred completely to storage first. After the complete transfer the AS55X reboots. If the transfer aborts, you can try again. The old firmware will be stored until the new firmware is finally loaded. For this reason it´s harmless to do a firmware update via LAN or GSM.

All active calls will be disconnected after the whole transfer.

There is one firmware file for all AS55X.

The current version of the AS55X you will find at https://www.kuhnt.com.

6.6 Download announcement file

Here you can download the announcements files to the AS55X. The selected announcement files will replace the current announcements.

Various announcement files (e.g. different languages) you will find at https://www.kuhnt.com.

If you would like to use some individual announcements, please contact your dealer.

6.7 Enter upgrade code

Your AS55X could be equipped with some additional features. To check which features are activated, look at services menu (see 7.15). To test these features you have the option to activate a 28 days test license (see 5.13).

If you want to activate these features you have to send the listed upgrade key to your local dealer. He will supply you with an upgrade code, which have to be entered in the upgrade code field. After pressing the "Apply" button, the additional feature will be available. If you press "define default" the activated feature will be available furthermore without entering the upgrade code again.

7   Services
7.1   Trace

The trace function is a basic service for troubleshooting. All connections and control data will be listed in the trace window. This is feasible for GSM, ISDN, SIP and LAN. It is possible to trace via local or remote access. But keep in mind, that your remote connection is also listed in the trace.

The option to save your current trace or load previous data, you can find under the "file" entry in the menu bar of the trace window. To load data, you have to switch your service software into "demo session" mode. It is also possible to export your trace data to a text file or to a pcap-wireshark file. For better trace results (timestamps) check the AS55X clock before starting the trace.

Under the menu bar you can find the record control array. You can choose between forward and backward trace. The forward trace is a real time trace that shows you the current data traffic. You can start and stop the trace with the buttons below. Press the "Pause" button to pause the trace temporarily. The shown trace data in the window will stop, but the AS55X will store all following information. This data will be shown after you press the "Pause" button again. Then the further trace will run normal. A backward trace will not trace in real time. The AS55X is equipped with a first-in first-out memory. If the memory is full, the oldest data will be overwritten. So you have the opportunity to get last trace data from the past events. All buffer data will be stored beyond a device reboot. So it is possible to edit the last backward data after an error occurred. The period of time depends on various setting (see 0).

Below the record control you can find some trace filters. Herewith it is possible to refine the results of the trace data. You can make a general selection between ISDN, LAN, GSM, control data and transaction numbers. A fine selection can be made in the menu bar at the point "Entry type filter". GSM data can be separated by all or by only one single channel. Finally you have the option to use a transaction numbers. Every call or call setup will be indexed with its own transaction number "TAN". For example you will locate all trace data to one phone call. If there is much traffic, you can search a transaction. Therefore press the "Find TAN"-button and type in the called phone number. All TANs with this number or this part of a number will be shown. These TANs are also listed in the exported Wireshark trace. Keep in mind that the timeline is not correct because these TANs and a consecutive number are inserted there.

The progress indicator for example is used if a long trace will be loaded and this import will take a few seconds. So it is also possible to use the "Stop"-button to abort a longer import. "Exit" will quit the trace window.

Finally you have the "Cell info"- and the "Show with Wireshark"-button in the menu bar. Here you have the option to open a single LAN messages in Wireshark. For all other messages this button is greyed out. The "Cell info" menu items are not greyed out, but there are still some adjustments needed to get GSM cell information. In the special settings menu (see 5.12.1) you have to increase the repetition timer. For normal use this data are not required, so that the default value is zero seconds.

On the right side you can find the major trace window with all trace data. First you have a consecutive number and a timestamp. The TAN will be assigned to every new call or call setup. In the next column are listed the sources of every entry. A short info you will find in the last column. After selecting one row you can see all data at the bottom of the trace window.

If you sent a trace file to your local distributer it is not necessary that you make a backup of your configuration. All device data are stored within the trace record data.

7.2   View GSM channel states

Here you get some information about your GSM channels.

First you can see how much channels are activated. Not activated channels are marked red (see 5.9.2). The Software detects as many channels as listed in menu "GSM Settings – General – Equipped GSM channels". When a GSM module is insert, it appears here on "Engine type" with its name, otherwise the module is damaged. The hardware and software version of each module is also listed. The driver hardware revision is the software release of the circuit board. These should be the same on each channel. The IMEI number of each GSM module is listed for maintenance purposes here.

The IMSI number of your SIM card is listed next. In the first five digits indicate your home network (MCC and MNC). The next eleven digits of the IMSI are your MSIN.

The flash error code indicates the same errors as the LED outside the AS55X. These errors are described in the following table.

The state shows you the condition of the GSM channel. It can be "registered", "unknown" or "SIM not ready". If you are registered to the wireless network, your signal quality is listed in the field below. For good voice quality, the value should not less than 40-50 percent. The network, listed with name, MCC and MNC, is shown in detail, just to check if you are logged in the right network.

Flash Code Error	LED indicator	ERROR
	continuous red light	
Engine out of order	1x red flashing	No module insert or module damaged
SIM missing	2x red flashing	No SIM insert or damaged
SIM error	3x red flashing	SIM unwilling, required no PIN or required PUK
PIN of SIM unknown	4x red flashing	SIM requires PIN but no PIN is entered in the Software
SIM of PIN wrong	5x red flashing	The entered PIN is wrong
Last Attempt of entering PIN	6x red flashing	3rd attempt of your PIN required, not possible with AS55X
No registration	7x red flashing	No registration possible
Failed to choose network	8x red flashing	Log in not possible, designated network not permitted
Data Exchange Error	10x red flashing	Errors in data exchange with the GSM module
Echo Command Error	11x red flashing	Error in the benefit of "echo cancelling level"
Reset by Timer	12x red flashing	Reset by Timer (GSM channel restart time)
Mux Start Error	13x red flashing	Error in the context of "Telnet mode"

7.3   Control GSM channel

Here you have the option to control the build-in GSM module by AT commands. Use the commands of the preset list or the AT command set from the Cinterion manual. Contact your local dealer to enable your AS55X-Service software to use this feature.

7.4   View number management

For troubleshooting/debugging, you can edit here a list of the last 30 calls. The table shows the incoming, dialled and outgoing number of every call.

First you see if the call is initiated on the wired network or started on GSM. All calls from GSM network are listed with the corresponding channel. Calls from the wired Network are listed with an internal task number between 1 and 30. The listed calling numbers are for editing how your provider or your PBX forwards the source calling number. If the AS55X detects a special type of number (e.g. international, national), it is listed in the field "source calling number type". All directly routed calls are listed in the column "Source called number". In the "destination location / channel" you can check if the outgoing channel is correct (e.g. if the "individual call setup" is OK?). All DTMF routed calls are listed in the next columns. You can check if the typed number is correct. Sometimes single numbers are not detected correctly or listed twice. Therefore you can check the "timeout DTMF recognition" in general GSM settings (see 5.9.1). The outgoing identity is listed in "destination calling number", the called number is in the last column.

7.5   View call setup list

This list is to monitor the callback function of the AS55X. In the menu "global call-setup" you can choose the call setup type. Thereafter every call or every unanswered call is shown is this list. The last 1000 calls are saved. You can see the number from wired network and the called mobile number. The function "delete call after callback" is also note down here. Finally the remaining time of every call is listed. After the expiry of this time the number is deleted from the list.

The registered numbers are managed independently from the cards. An outgoing call via module 1 can be answered via module 2. There is only an n adjustment on the numbers.

The open list can be updated with the "Refresh" Button.

7.6   View ARP table

In the ARP table all MAC addresses are stored, which are used by the gateway. It is only a conversion list from MAC addresses to IP addresses. Normally, only the addresses of your PBX, of your default gateway or of your service PC should be listed here. If a MAC address is not longer in use, it will be deleted automatically after one hour.

7.7   View LAN status

In the menu "LAN Status" you can check its network configuration. First, you can detect whether the network interface is enabled. If this is active, the right LED lights on the RJ45 jack. You can also see is if the physical network interface settings are correct. IP address, subnet mask, default gateway and DNS servers should be listed as entered. The IP address of the STUN server and the public IP will be shown only if a STUN server is configured.

Here your MAC address will appear with its current configuration. If you change it at the menu "Special setting", it will be shown here. In the menu "View device data" the factory-set MAC address is shown, which is set in "Define default".

With the "Sent PING to…" button you can check if the configuration is OK. If the destination response to ICMP messages, you will see the response time. Otherwise you will get a negative response within four seconds.

7.8   View SIP status

Here you can basically see SIP information, particularly if you configure your AS55X in SIP client or SIP peer mode (for SIP Server see 7.9). First you can see if the SIP mode is configured correctly. If the SIP mode is set to peer or client and the connection is correct, the state must change to "Ready". Additionally in SIP client mode the registrar IP address is also listed. If no IP address is listed or the state keeps "Not ready", there will be a failure in configuration or installation (e.g. active ISDN mode, DNS translation problems, bad gateway or bad cable connection). If you configure SIP peer mode, the IP address of the registrar keeps "0.0.0.0" and the 1st or 2nd peer detection changes from "Undefined" to "YES". In SIP client mode there will be one or more extra column. You can see the state of your SIP extension. In the line "Register status of" you can see if your login was successful. If the registration was correct, the AS55X will confirm with "Registered". Otherwise you will find the response code (e.g. registration denied or response code 404).

7.9   View SIP registered user list

In this menu all registered clients will be listed. They are shown with additional information. From the REGISTER message the clients IP address, the SIP port and finally the expire time will be saved and listed here.

If no user is listed, check also the SIP state in "view SIP status" (see 7.8). If the SIP status is ready, check your telephone configuration or check the authorized user list (see 5.7.4).

7.10 View statistics

In the statistic menu, you can see all data which are captured by the AS55X. First you can see the runtime of your system. This is the power-on time and not the time since the first start. The duration of this statistic is also the power-on time since the last cancelation or since the first boot. In the table below, you can see various minute counters for GSM calls and sent short messages and the signal quality statistics for each channel of your gateway. Also in this time, the ISDN layer 2 errors will be logged, the reboots will be counted and the minimum and maximum temperature of this device will be monitored. The connected minutes are

displayed exactly but in full minute steps. E.g. three connections with each 15 seconds connection time are shown as 0 minutes. All displayed values are real-time data and will refresh automatically in local and LAN access. If you use GSM data exchange use the "Refresh" button to update all displayed data.

This statistics will not be deleted after a firmware update. Non active channels are highlighted red.

7.11 Restarts

Here you have the option to restart parts of the AS55X or the complete device. First you have the option to restart all GSM channel. If you use GSM data exchange, you would cut your own connection. For this reason it´s also possible to restart only one of the GSM channel.

Finally you have the two options "restart cold" and "restart warm". Both will restart the whole device. If you use the "warm" mode, only a few device data will be deleted. This includes the list "number management" and the "ARP table" (except SIP Server mode). In addition to the "warm" mode", the "cold" mode will delete every list. This includes also the "number management", the "ARP table" and the "call-setup list". In both cases the statistic and the statistics will not be deleted.

7.12 System clock / calendar

Here you have to set the internal clock. It is mandatory for a correct function of the SimSwitch. But it´s also recommended for other functions (e.g. correct trace data). For more accurate time settings within your AS55X it´s possible to enable the Daylight Saving Time rule. Then the AS55X adjusts the time changes by itself.

To set the time in AS55X, it is also possible to import from your computer clock. Therefore press the "Import computer clock" button on the right side. In some older firmware releases the correct day must be set manually.

7.13 View SimSwitch status

Here you can check you configuration and the minute counters of your AS550 or AS551 SimSwitch. Therefore you have to configure the AS55X in SimSwitch mode (see 0). All channels which are configured as SimSwitch will be listed here. On the top you can choose this configured channel.

You can see how much SIMs are configured for each channel and how many minutes or short messages are used with each of these SIMs. You can also reset the minute counter or switch to one special SIM manually. You can identify the currently active SIM by a look at the "Active" column. The active SIM is marked. The listed minutes are displayed in real-time.

Below the listed minutes, you can see how your abort causes are configured. Finally you have the option to see the statistic of the last three time slots of each channel.

7.14 View device data

The device data menu is an overview of all basic device data. Such as the device type, the serial number and the initial date of the manufacturer, MAC address, firmware release etc.

The details are necessary for support enquiries. For any problems with your AS55X, please note this information. First you have hardware, firmware and boot switch revision. These are necessary to know which hardware and firmware is used. You can also look up this information to evaluate if new firmware is available. New releases you will find at https://www.kuhnt.com. They will contain bug fixes and sometimes new features. If you want to downgrade your devise you can check the first supported firmware revision for this device ("Minimum firmware revision"). Older firmware releases will not be accepted by the AS55X.

Finally you can find your device name, its temperature and the MAC address. The value of the temperature is readout only at the beginning of a programming session. It is no realtime indication. Furthermore the polling of this sensor is made first about one minute after the device has started. So it can happen that no temperature is listed if you start your session near after a restart of the devise. The listed MAC address is the default address. If you change the MAC address in the menu "Special setting", the current setting will be shown in the "view LAN status" menu (see 7.7).

7.15 View extended features

Look here for an overview about your AS55X options. All activated extended features are marked in this list. The amount of available Telnet channels is also listed.

If you use an upgrade key, the activated feature will be displayed as marked. After a firmware upgrade (see 6.5) or a reset to factory default ("define default") all extended features will persists.

7.16 View test licenses

This shows the state of your test license. You can see if the test license is running or stopped. The remaining time persists after a firmware update and stops if the AS55X is powered off. For more details see 5.13.

7.17 Short Message Control

This "short message control" is designed for developers to assist them in development of new SMS applications and includes all important messages which can be sent by the application to the AS55X. First you have to enter server IP address and port. If you now press "Connect to server" a new telnet session to AS55X will be established. If you now want to test sending or receiving a short messages you can use the templates button on the right. If you press one of these buttons a crude structure of a message will be shown in the upper textbox. You now have to add channel number, message ID and phone number (in international form). Finally you have to press "SendCommand" to transmit this request to

your AS55X. All received replies will be shown in the lower textbox. To finish your session press "Disconnect from server".

	By default this feature is not shown. If you want to test SMS Server or to create a software application, please contact your local dealer to enable this menu item.

8   Release Notes

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 AS55X Service Software. This firmware can be used for all supported AS55X hardware versions.

8.1   Release 3.0

8.1.1   New and improved features

- 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'.

- Additionally add prefix '0' to CLIP to wired interface

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

8.1.2   Problem solution

- 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.

- 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.

8.2   Release 3.01

8.2.1   New and improved features

- 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.

- 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.

- DialCommand

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

8.2.2   Problem solution

- 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.

- 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.

8.3 Release 3.03

8.3.1   New and improved features

- 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.

- 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.

- Sent short messages in statistics

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

- 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.

- Get calling party number from Remote-Party-ID header

  The default search sequence for determining the calling party number from SIP headers is now:
  o P-Preferred-Identity
  o P-Asserted-Identity
  o Remote-Party-ID
  o From
  o Contact

- SIP UPDATE implemented

8.3.2   Problem solution

- 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.

- 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.

- 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.

- 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.

- 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.

8.4 Release 3.04

8.4.1 New and improved features

- 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.

- 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.

- 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.

- 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.

- 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.

- Receive dialable GSM numbers in SIP client mode

  The AS55X in client mode can now receive a dialable 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 dialable GSM number.

8.4.2   Problem solution

- 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.

- 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.

8.5   Release 3.05

8.5.1   New and improved features

- 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.

- 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.

- 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.

- 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.

8.5.2   Problem solution

- 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).

- 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.

- 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.

8.6   Release 3.10

8.6.1   New and improved features

- BRI interface control

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

- 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 manger, can be found in the subfolder 'Documents' of the service software.

- 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.

- 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.

- 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.

- 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.

- 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.

8.7   Release 3.11

8.7.1   New and improved features

- WLL control

  From this firmware, the control of the new WLL 550 PRO is implemented.

- 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.

- 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.

8.7.2   Problem solution

- 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.

- 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 an interface reset of the AS55X.

8.8   Release 3.12

8.8.1   New and improved features

- 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.

- 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 call.

- Add a leading '0' to the dialled 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.

- 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'.

- 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'.

- 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.

- 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.

- 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. Those are mostly calls from the wired network to an extension that is forwarded to a GSM number (via the AS55X).

- PassThrough backup

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

- 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 are not available with EU-3 !

- 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.

- Dedicated network type for EU-3

  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.

8.8.2   Problem solution

- WLL: Immediate work after device reset

    After a device reset with the reset button it is no longer necessary to download 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.

- 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.

- 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.

8.9 Release 3.13

8.9.1   New and improved features

- Path replacement

    Path replacement for ISDN QSIG applications is implemented, please ask the customer support for configuration.

- Use any channel if selected group not ready

    If for an outgoing call to GSM all channels of the by routing selected GSM group are busy or not registered, this call can be handled by any other available GSM channel. This feature has to be configured in 'Global call-setup'.

- DHCP active after reset

    After clearing configuration with the reset button, the internal DHCP server is now active. Hence, after reset the AS55X can also be accessed from computers without fixed IP addresses and without any additional settings to their network adapters if no other DHCP server is available in the network.

- PH8 supported

The firmware now supports the Gemalto/Cinterion PH8 module. With the WLL550, download speed of up to 7.2Mbps is possible.

8.9.2   Problem solution

- Fixed link ISDN B-channel to GSM channel

  This issue only concerns the AS551 BRI 4V. If the fixed link between ISDN B-channel and the GSM channel is activated, this did not work properly for the GSM channel 3 and 4 in the direction from GSM to ISDN. This problem is solved.

- Character '@' with SMS API

  The character '@' (code 0 in the GSM alphabet) could not be sent or received via the SMS API. This problem is solved.

- SIM blocked

  Some SIM types have very long startup delays and during this delay, the SIM seems to be blocked. The firmware can handle this issue now.

- Suppress SIP protocol elements

  Some PBXes cannot handle the SIP PROGRESS request or session description (SDP) in the SIP RINGING request. PROGRESS can be suppressed by setting unnamed bit 68 and SDP in RINGING can be suppressed by setting unnamed bit 69.

- Authentication to SIP BYE

  Some PBXes need an additional authentication for the termination of a call. The AS55X can handle it now.

- DNS and DHCP with WLL

  In some cases DNS requests would not be forwarded to internet properly if no explicit DNS server IP address has been configured in WLL550 (e.g. if the computer gets its IP address from the WLL550 with DHCP). This problem is solved.

8.10 Release 3.14

8.10.1 New and improved features

- Routing by source numbers

  Now it is possible to select a GSM channel group according to the CLIP of the wired side. Hence, GSM channels can be assigned to extensions. In PassThrough mode, this feature also can be used to avoid interception of calls from single extensions of the PBX.
  In the individual call-setup list, the domian of the number (GSM or PBX) can be chosen. PBX numbers always have higher priority and are placed onto the top of the list. The routing information will be taken from the first matching entry in this list. E.g. if for a specific call both the extension number and the GSM number exist, the routing command of the extension entry will be used.

- ProviderSimSwitch

  The AS550 WLL now can be shipped as SimSwitch. SimSwitch in this case is not primarily designed for saving of costs. It is rather meant for looking for the best supply by any provider. This feature has to be licenced and configured. It is possible to enter an internet access set for each GSM/UMTS-channel. There are some commands to be given from the wired side, that can control this facility.

8.10.2 Problem solution

- Noise on audio calls with PH8

  An audio call over the PH8 GSM/UMTS module can result into loud noise in single or both directions. This problem is solved.

8.11 Release 3.15

8.11.1 New and improved features

- Trace forward during LAN service session

  If all received and sent LAN frames shall be traced, the frames for the service channel between computer and AS55X are excluded. Hence, trace forward in this case is possible now.

- New module release of TC63i

  The TC63i release 02.004 is supported now.

- SIP upgrade

  AS550 and AS551 BRI can now be upgraded for use with SIP.

- Reverse IP remote access

  Especially for the AS550 WLL, but in few cases for all other AS55X as well, there are Situations that prohibit from access via the existing remote access channels. Hence, a new channel is implemented so called 'Reverse IP access'. For more information ask the manufacturer.

- Remote access via Telnet channel of AS55X

  An AS55X with Telnet can now directly and without a virtual COM port be used as hardware for a remote access session via GSM data. For use of this feature, Telnet must be selected instead of a COM port with the 'Via' selector.

- Optimisation of the FAX stack

  There are optimisations of the FAX stack done. Especially an intermediate quality check is ought to increase the probability of a successful transmission.

8.11.2 Problem solution

- Short message sending problem with MC55

  With MC55 GSM modules the sent short message could be corrupted. This problem is solved.

- ISDN Sending complete in ISDN

  Receiving the Sending Complete information element in ISDN will now override the expectation of further suffix digits in PassThrough mode. This is for better interception of calls with different called number lengths.

- Park extension in SIP

  Some PBXs assume an error if it parks a SIP connection and the peer entity does not stop sending the audio stream immediately. The AS55X takes account of now.

- WLL550, Telephone rings instead of FAX device

  In some cases if the calling FAX device does not send the CLIP to the WLL550 via GSM, the call was not assumed correctly as FAX call and

the telephone was ringing. This problem is solved.

- Charge signal only for outgoing calls

  The charge signal for AS55X PRI and BRI was generated for outgoing calls to and incoming calls from GSM. This is corrected and form now on only present for outgoing calls to GSM.

- BRI synchronisation problem

  In single cases, call activities on that line, used for Synchronisation of the BRI interfaces could disturb the normal operation of the AS55X. This problem is solved.

- Blocking of BRI interface

  There can occur an error situation, triggered by data communication errors between PBX and AS55X BRI, that make the AS55X remind in a permanent error state and block all calls. This problem is solved.

- GSM netcode error

  With Firmware version 3.14 the GSM netcode did not work in every cases, this is corrected.

8.12 Release 3.16

8.12.1 New and improved features

- Restrict speech codecs

  In some areas there is audio distortion based on the usage of a specific codec in GSM network. With this firmware release, the allowed codecs can be restricted by settings of unnamed byte 122.
  0: All codecs allowed
  hexadecimal 81: AMR disabled
  hexadecimal 82: HR disabled
  hexadecimal 80: AMR and HR disabled

- Star hash codes enabled for WLL

  It is possible now to enter start hash codes e.g. to switch call forwarding in gsm networks. All codes beginning with a star can be entered normally. But all codes beginning with a hash must be entered in a special manner as the beginning with a hash will result in a conflict with DTMF commands of the WLL. For codes beginning with one hash please enter '1#', for codes starting with a double hash please enter '2#'. E.g. to switch call forwardings off, '2#002#' has to be entered. To enable this feature, unnamed byte 9 must be set to 0 and unnamed bit 75 must be set.

- CLIR with FAX calls

  Outgoing FAX calls present the voice number instead of the FAX number of the SIM. This can result into confusion by the recipient, which number should be used for a reply. If unnamed bit 74 is set, the presentation of the number will be restricted, even if it is requested for voice calls.

- Support of firmware version 03.001 of PH8 module

- Simplified internet access of the WLL550

  As long as internet access is not set by service software, the access can be enabled by the DTMF code #721# and disabled by the code #720#. This setting will be removed with a cold start.

- Support of WLL550 PCB revision 2

8.12.2 Problem solution

- Suppression of FAX error correction

  If the receiving FAX device provides polling, the suppression of error correction by the WLL550 did not work properly. This is corrected.

- Improvement of the DHCP server

  For WLL550 the DHCP server is improved for better handling of a changed IP address by the wireless network.

- Very long boot time of GSM module

  In some cases the GSM module needs very long time to start with reaction of commands and later to return the IMSI of the SIM. This is due to handling much data from the SIM. This will be tolerated now.

- SMS callback request and VPN prefix

If the VPN prefix was used, SMS callback request did not work. This problem is solved.

- SMS callback request with calling number missing

  If the short message callback request was activated, the placeholder for the calling number was used and if the calling number actually was not present, the device could perform a restart. This problem is solved.

8.13 Release 3.17

8.13.1 New and improved features

- Support of PLS8 The firmware now supports the PLS8 module.

  With the AS550 WLL, download is possible with LTE speed.

- APN list enhanced For dialing in to wireless internet, the APN list contains the following entries (netcode, APN, username, password):

  26201,internet.t-d1.de,td1,gprs
  26202,web.vodafone.de,-,-
  26203,internet.eplus.de,eplus,gprs
  26207,internet,,
  20404,web.vodafone.nl,vodafone,vodafone
  20408,internet,KPN,gprs
  20412,internet,,
  20420,internet,t-mobile,t-mobile
  20601,internet.proximus.be,-,-
  20610,internet.be,mobistar,mobistar
  22801,gprs.swisscom.ch,gprs,gprs
  22802,internet,internet,internet
  22803,internet,,
  23201,a1.net,ppp@a1plus.at,ppp
  23203,gprsinternet,gprs,
  23205,web.one.at,web,web
  23207,web,web@telering.at,web

  If this list ought to be used, checking of 'Go online' in 'Wireless internet access' or entering the DTMF code '#721#' will be enough. If another or an alternative data set is necessary, it can be entered via 'Wireless internet access'. For Provider SimSwitch, please enter a '?' as APN in order to use this list.

- SIP instant messaging

  For sending of short messages, the SIP method MESSAGE is implemented. By default, the AS55X immediately answers with 202 final response (Accepted) after it received and accepted a message request. This behaviour can be changed with the unknown byte 127. With value 1, the AS55X will first send a provisional response and a final response, when the short message sending is completed. With value 2, the provisional response will be omitted.

8.13.2 Problem solution

- FAX call handshake problem if sending FAX has no sender identification In this case the handshake failed.

    This problem is solved.

8.14 Release 3.19

8.14.1 New and improved features

- Redesign and enhancements of short message features

    Sending and receiving of short messages is possible now via a web server over all GSM channels of the AS55X. A full conversion of the character tables from/to GSM is implemented. With the configuration, a remote IP address can be linked to a dedicated GSM channel of the AS55X or a login with a set of username and password can be used to address a GSM channel. Either a short message handling parallel or concurrently to voice calls can be configured.
    Sending and receiving of short messages with more than 160 characters is possible now via the SMS API as well as using of the alphabet conversion from/to GSM.
    Short message callback can also be used now with the WLL550.

- Support of PLS8 Version 2

    This firmware supports the PLS8 module version 2 with voice calls and short message services.

8.14.2 Problem solution

- Some smaller issues solved.

8.15 Release 3.20

8.15.1 New and improved features

- Internet access data list

  Additionally to the hardcoded APN list there is a configurable one. This list is useful in cases, the SIM could be changed during operation and the provider could not be located in the hardcoded list. Furthermore there is a switch to allow or prevent data roaming.
  Now there are four input fields where internet access data can be entered. If provider SimSwitch is activated, the entry according to the SIMs has the highest priority. The entry in wireless internet access screen has the next priority level followed by this new internet access data list. The hardcoded list got the lowest priority level. And if no matching data set could be found at all, internet access without APN will be tried.

- Support of BGS2

  The firmware now supports the BGS2 module within the AS550 BRI.

- SimSwitch Optimisation

  Provider SimSwitch has been optimised especially for the usage in different countries.

- DTMF code to allow data roaming

  The DTMF code of the WLL550 for internet access has been enhanced to allow data roaming. Use the '#723#' for internet access with data roaming. The already known code '#721#' keeps its function, internet access without data roaming. Both codes are irrelevant, if internet access is configured with the service software.
  Please keep in mind, your SIM contract must permit data roaming as well!

- SNMP enhancements

  The mobile network type, if GSM, EDGE, UMTS or LTE, can now be requested via SNMP. Until now the GET-NEXT requests within the table firstly increment the table element and when the whole table is processed, the index will be incremented. This order can be changed by setting the unnamed bit 45.

8.15.2 Problem solution

- FAX number not displayed by default

    As GSM networks always display the voice number with outgoing FAX calls and a FAX-callback to this number must fail, this number is now suppressed by default. By clearing the unnamed bit 74, this suppression stops.

- Redesign of operator selection

    With some cellular modules, changing of the operator selection is a time consuming process. This lengthens startup duration and causes problems in conjunction with the searching for the best network of provider SimSwitch with different operator selections.
    Operator selection will be performed no longer. Now it will be checked if the module is registered to the requested network. In exceptional cases where the requested network is not the home network to the SIM, this process could fail. With setting of the unnamed bit 82 the former behaviour will take place again.

- SIP problem with authentication

    If the AS55X is configured as SIP server with authentication (entries in the authorized client list with passwords) and if the SIP client answers to the authentication request from the server within microseconds, this can result into a rejection by the server. This problem is solved.

8.16 Release 3.21

8.16.1 New and improved features

- SIP with WLL550

  There is an upgrade available for SIP over the WLL550..

- FAX over SIP

  There is an upgrade available for FAX over SIP. This one includes WLL550 and all AS55X with SIP or SIP upgrade. One FAX can be handled at time, but over any GSM channel. One or more extensions can be configured as FAX terminal.

- SIP instant messaging enhancement

  This feature is enhanced for multipart short messages of up to 459 characters.

- PLS8 version 3 supported

  The firmware now supports the PL8 module with version 3.

- APN selection based on IMSI

  Until now the APN selection was made based on the network, the channel is registered at. With this firmware release, the default base for the selection has been changed to the netcode of the SIM IMSI. With unnamed bit 83 set, the former behavior will take place again.

8.16.2 Problem solution

- Restart after CRC4 mode changed

  If the CRC4 mode of the running ISDN PRI interface is changed, a device restart will be performed in order to make the setting effective immediately.

- Redesign of operator selection

  With some cellular modules, changing of the operator selection is a time consuming process.
  This lengthens startup duration and causes problems in conjunction with the searching for the best network of provider SimSwitch with different operator selections.
  Operator selection will be performed no longer. Now it will be checked if the module is registered to the requested network. In exceptional cases where the requested network is not the home network to the SIM, this process could fail. With setting of the unnamed bit 82 the former behaviour will take place again.

- Display of CLIP with WLL550

  If the WLL550 is not programmed, the Calling party number will now be displayed correctly with the international prefix.

8.17 Release 3.22

8.17.1 New and improved features

- A

  A

- B

  B

- C

  C

8.17.2 Problem solution

- D

  D

- E

  E

- F

  F

Personal memo: