

Aarenet System and IPPBX settings to connect Yeastar MyPBX U100 IP-PBX through SIP Trunk

Revision History

Autor	Date	Info	Version
F.Remmers	12.09.2017	Initial Version	1.0

All information without warranty and valid only based on the used hardware and software and only for informative use. Settings can be different on different Aarenet systems.

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Aarenet advised base settings

Settings can be system specific and/or SIP phone specific. If deviating values are used, connection issues may result. All information informative and without warranty. The settings can be different for different Aarenet systems. Basic setup of a SIP Trunk on the Aarenet system is not part of this document.

Advised system / SIP phone settings:

- Voice Codec: G711A
- FAX Codec: G711A
- Hold Methode: direction attribute „sendonly“
- DTMF Methode: RFC 2833
- Session Timer: 1760s
- SIP Registration Expiry: 300s

Aarenet SIP Trunk Config Center Settings

The image shows four screenshots from the SIP Trunk Config Center interface:

- Account Settings (Top Left):** Shows account details for 'aan1-000088'. The 'Addresses' field is circled in red and labeled '3', indicating 10 addresses. The 'Username' field is circled in red and labeled '2', with a note 'DifficultUsername'.
- Address Configuration (Top Right):** Shows configuration for address '0449980510'. The 'Domain' field is circled in red and labeled '1', containing the IP address '213.173.185.12'.
- Addresses of Account (Bottom Left):** A table listing 10 addresses for account 'aan1-000088'. The first address, '0449980510', is circled in red and labeled '4'.
- Registration Status (Bottom Right):** Shows registration details for '0449980510@213.173.185.12'. The status is 'registered' (with a green dot), circled in red and labeled '5'.

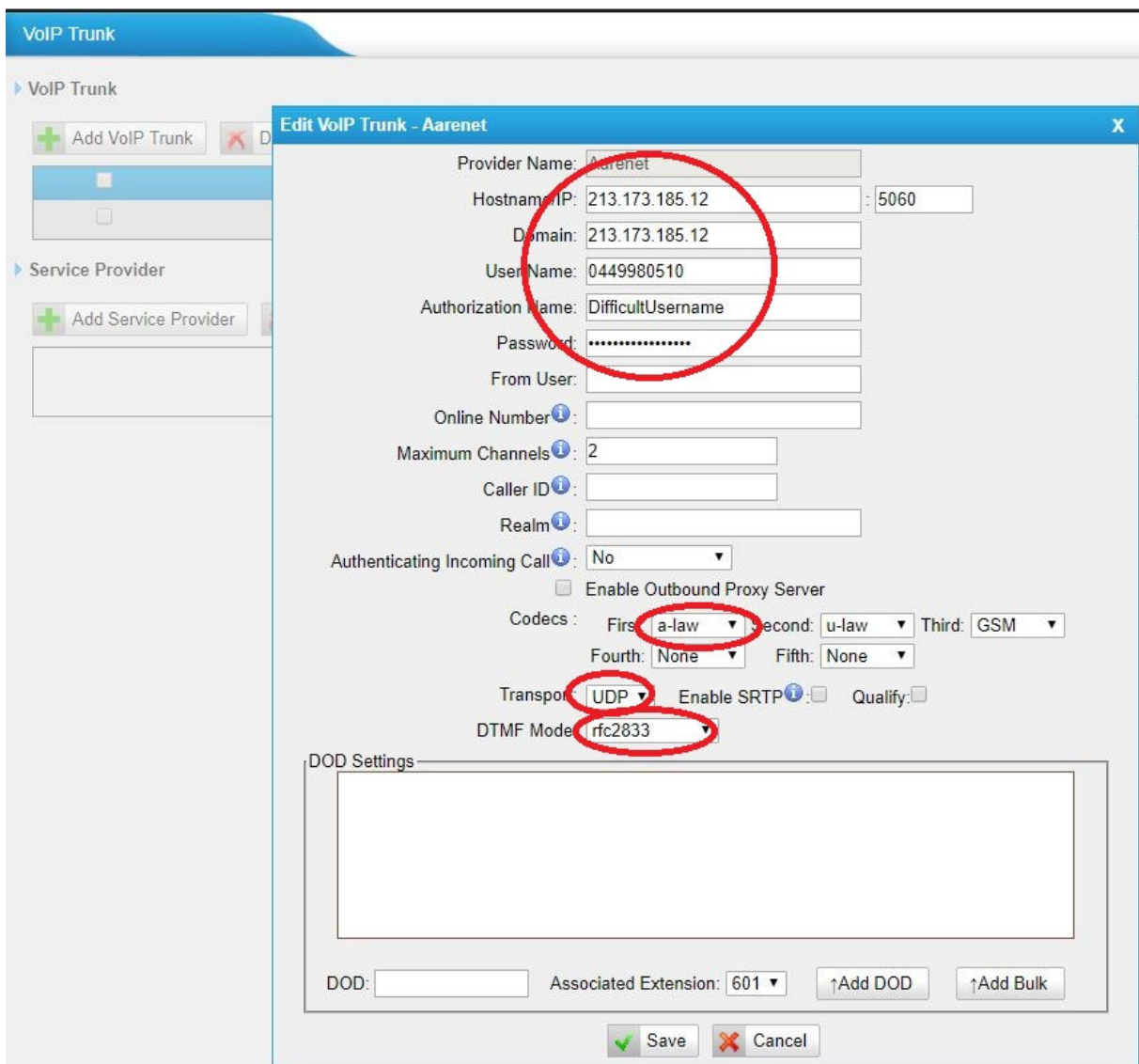
1. IP Adress for SIP registration
2. Username / Password info
3. Info 10 DDI Number
4. List with single DDI
5. Confirmation registration Yeastar MyPBX on SIP Trunk

Yeastar MyPBX U100 direct SIP

Setup	MyPBX direct on Aarenet SIP Trunk
Devices in use	Siemens 2010, Yealink T46G
External	079xxx6113, 079xxx4594
Tested	
Incoming calls	OK
Incoming calls with CLIR	OK
Outgoing calls	OK
Internal calls	OK
User Groups	OK
Fax	Not tested
DECT	Not tested
Registration	OK
Session Refresh timer	OK default 300 sec.
DTMF outgoing	OK
Early media to mobile (disconnected)	OK
CF external to internal	OK, using usergroups and «announce»
Call to 161	OK
Call to 0900 55 33 11	OK
3 party conference	OK
Transfer with announcement	OK
Transfer without announcement	OK
Swap calls	OK

Important: Please make sure that no ALG is activated in the pbx nor in the NAT Router.

Step 1. Add SIP Trunk



Make sure the credentials from the config center are correct. Select correct codec order, Transport protocol and DTMF mode.

Status	Signal	Trunk Name	Type	User Name	Port/Hostname/IP
Registered		Aarenet	SIP	0449980510	213.173.185.12
Disconnected		pstn1	FXO		Port 1
Disconnected		pstn2	FXO		Port 2
Disconnected		pstn3	FXO		Port 3
Disconnected		pstn4	FXO		Port 4

The Status overview shows an active SIP Trunk on the Aarenet system

Step 2. Allocate DDI Nummern to subscriber

This is done through the programming of inbound routes.

Route Name	DID Number	Caller ID Number
TrunkMain10	0449980510	
TrunkDDI11	0449980511	
TrunkDDI12	0449980512	

Below Route (TrunkMain10) routes the number 0449980510 from Aarenet SIP trunk to the internal extension 601

Inbound Routes

[+ Add Inbound Route](#) [X Delete the selected Route](#)

Route Name	DID Number	Caller ID Number
TrunkMain10	0449980510	
TrunkDDI11	0449980511	
TrunkDDI12	0449980512	

Edit Inbound Route: TrunkMain10

General

Route Name: TrunkMain10

DID Number: 0449980510

Extension: 601

Caller ID Number:

Distinctive Ringtone:

Enable Callback: No [Callback Settings](#)

Member Trunks

Available Trunks	Selected
pstn1(FXO) pstn2(FXO) pstn3(FXO) pstn4(FXO)	Aarenet(SIP)

Business Days

Step 3. Outgoing traffic – allocate trunk /outbound rules

The first screenshot shows the 'Outbound Routes' page with a table containing one entry:

Route Name	Dial Pattern
ExternalTrunk	0.

The second screenshot shows the 'Edit Outbound Route - ExternalTrunk' dialog box. Key settings include:

- Route Name: ExternalTrunk
- Password: [Empty]
- T.38 Support: No
- Rmemory Hunt: No
- Office Hours: [Empty]

The 'Dial Patterns' section shows a table with one entry:

Dial Pattern	Strip	Prepend
0. 1		

The 'Member Extensions' section shows a list of available extensions and a list of selected extensions:

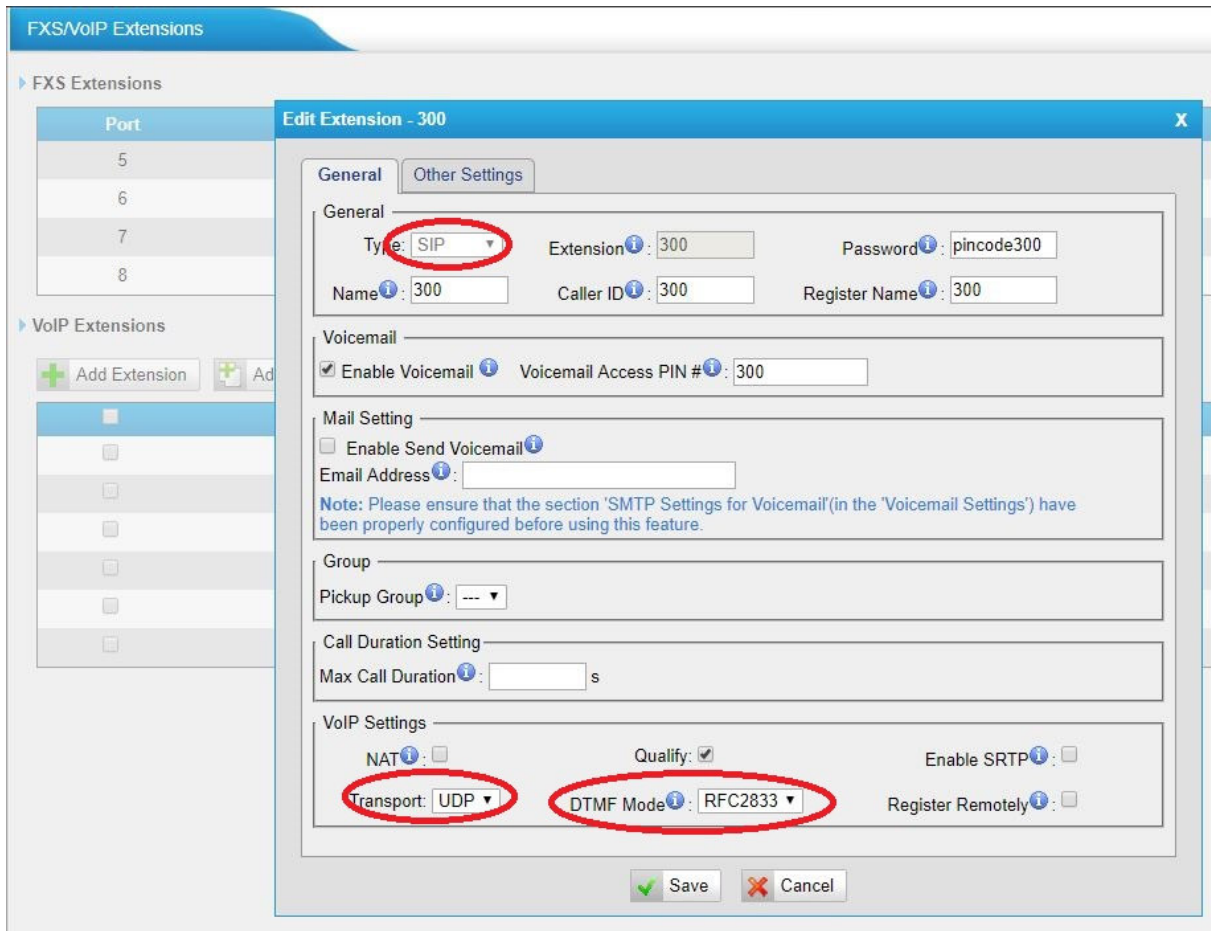
Available Extensions	Selected
[Empty]	300(SIP) 301(SIP) 302(SIP) 303(SIP) 304(SIP) 305(SIP) 601(FXS) 602(FXS)

The 'Member Trunks' section shows a list of available trunks and a list of selected trunks:

Available Trunks	Selected
pstn1(FXO) pstn2(FXO) pstn3(FXO) pstn4(FXO)	Aarenet(SIP)

Example: all calls starting with 0, made with one off the allocated devices are routed through Aarenet(SIP) trunk.

Step 4. Codecs and DTMF on device level



The screenshot shows the 'Edit Extension - 300' configuration window. The 'General' tab is active. The 'General' section includes: Type: SIP (circled in red), Extension: 300, Password: pincode300, Name: 300, Caller ID: 300, and Register Name: 300. The 'Voicemail' section has 'Enable Voicemail' checked and Voicemail Access PIN #: 300. The 'Mail Setting' section has 'Enable Send Voicemail' unchecked and an empty Email Address field. A note below states: 'Note: Please ensure that the section 'SMTP Settings for Voicemail' (in the 'Voicemail Settings') have been properly configured before using this feature.' The 'Group' section has 'Pickup Group' set to '---'. The 'Call Duration Setting' section has 'Max Call Duration' set to 's'. The 'VoIP Settings' section includes: 'NAT' unchecked, 'Qualify' checked, 'Enable SRTP' unchecked, 'Register Remotely' unchecked, 'Transport: UDP' (circled in red), and 'DTMF Mode: RFC2833' (circled in red). At the bottom are 'Save' and 'Cancel' buttons.

Please make sure that all relevant settings are set on each single device. Settings may deviate per device type and manufacturer.

Step 5. SIP Registration expiry

No information on SIP registration expiry settings found. The basic settings did not affect the performed tests.