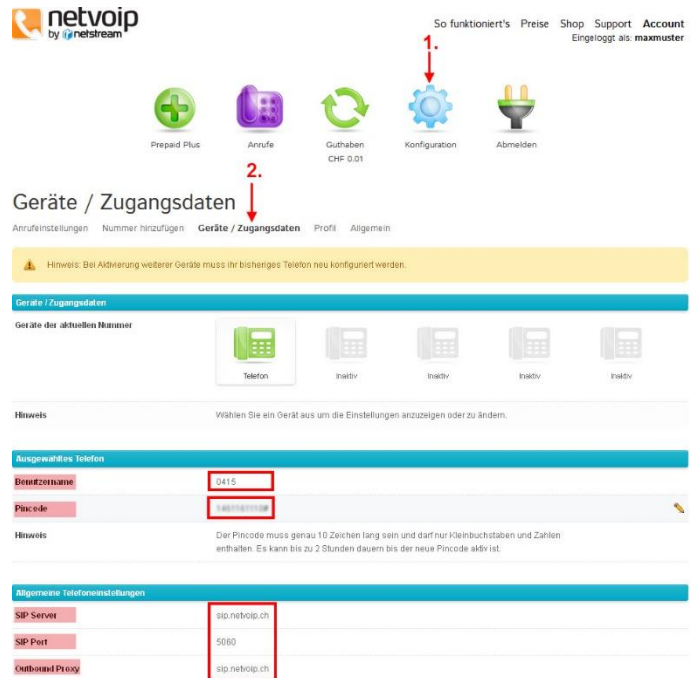


Bitte folgen Sie dieser Anleitung um Ihren Cisco SPA-122 mit Ihrem netvoip Account zu konfigurieren.

→ Schritt 1 Benötigte Daten

Für die Konfiguration relevante Informationen finden Sie in Ihrem netvoip Account auf netvoip.ch unter „Konfiguration → Geräte / Zugangsdaten“.



→ Schritt 2 Cisco SPA-122 Webinterface und Login

Verbinden Sie das Gerät gemäss der Cisco SPA-122 Bedienungsanleitung mit Ihrem Netzwerk. Die Standardkonfiguration des Cisco SPA-122 lautet wie folgt:
IP-Adresse: **192.168.1.15**
Benutzername: **admin**
Passwort: **admin**

Es wird dringend empfohlen das Standard Login Passwort zu ändern.



→ Schritt 3 Firmware

Diese Anleitung basiert auf **Firmware Version 1.3.1(003)**.



The screenshot shows the 'System Information' page in the Cisco Phone Adapter Configuration Utility. The 'Firmware Version' is highlighted in red and listed as 1.3.1(003) Dec 17 2012.

System Information	Value
Name	
Model:	SPA122, LAN, 2 FXS
Hardware Version:	1.0.0
Boot Version:	1.0.1 (Oct 6 2011 - 20:04:00)
Firmware Version:	1.3.1 (003) Dec 17 2012
Recovery Firmware:	1.0.2 (001)
WAN MAC Address:	64:9E:F3:C1:8D:38
Host Name:	SPA122
Domain Name:	(none)
Serial Number:	CBT160301YR

Die Firmware Version **1.4.1** finden Sie unter folgendem Link:

<http://www.cisco.com/c/en/us/support/unified-communications/spa122-ata-router/model.html#~tab-downloads>

Bitte laden Sie diese falls benötigt herunter und laden die BIN Datei im Menu „Administration → Firmware Upgrade“ auf das Gerät.



The screenshot shows the 'Firmware Upgrade' page in the Cisco Phone Adapter Configuration Utility. The 'Firmware Upgrade' menu item is highlighted in the left sidebar. The main content area shows the current firmware version as 1.3.1(003) and a file selection box for 'SPA122_S..._003.bin'.

➔ **Schritt 4.1**
SIP Einstellungen Teil 1
(SIP Parameters, SIP Timer Values)

Passen Sie die Konfiguration unter „Voice → SIP“ gemäss der Abbildung rechts an.

Quick Setup	Voice	Administration	Status
Information System SIP			
SIP Parameters			
Max Forward:	70	Max Redirection:	5
Max Auth:	2	SIP User Agent Name:	SVERSION
SIP Server Name:	SVERSION	SIP Reg User Agent Name:	
SIP Accept Language:		DTMF Relay MIME Type:	application/dtmf-relay
Hook Flash MIME Type:	application/hook-flash	Remove Last Reg:	no
Use Compact Header:	no	Escape Display Name:	no
RFC 2543 Call Hold:	yes	Mark All AVT Packets:	yes
AVT Packet Size:	ptime	SIP TCP Port Min:	5060
SIP TCP Port Max:	5080	CTI Enable:	no
Keep Referee When REFER Failed:	no	Caller ID Header:	PAD-RPID-FROM
SIP Timer Values (sec)			
SIP T1:	.5	SIP T2:	4
SIP T4:	5	SIP Timer B:	32
SIP Timer F:	16	SIP Timer H:	32
SIP Timer D:	32	SIP Timer J:	32
INVITE Expires:	240	ReINVITE Expires:	30
Reg Min Expires:	1	Reg Max Expires:	7200
Reg Retry Intvl:	30	Reg Retry Long Intvl:	1200
Reg Retry Random Delay:	0	Reg Retry Long Random Delay:	0
Reg Retry Intvl Cap:	0		

➔ **Schritt 4.2**
SIP Einstellungen Teil 2
(Response Status Code Handling, RTP Parameters, SPD Payload Types)

Passen Sie die Konfiguration unter „Voice → SIP“ gemäss der Abbildung rechts an.

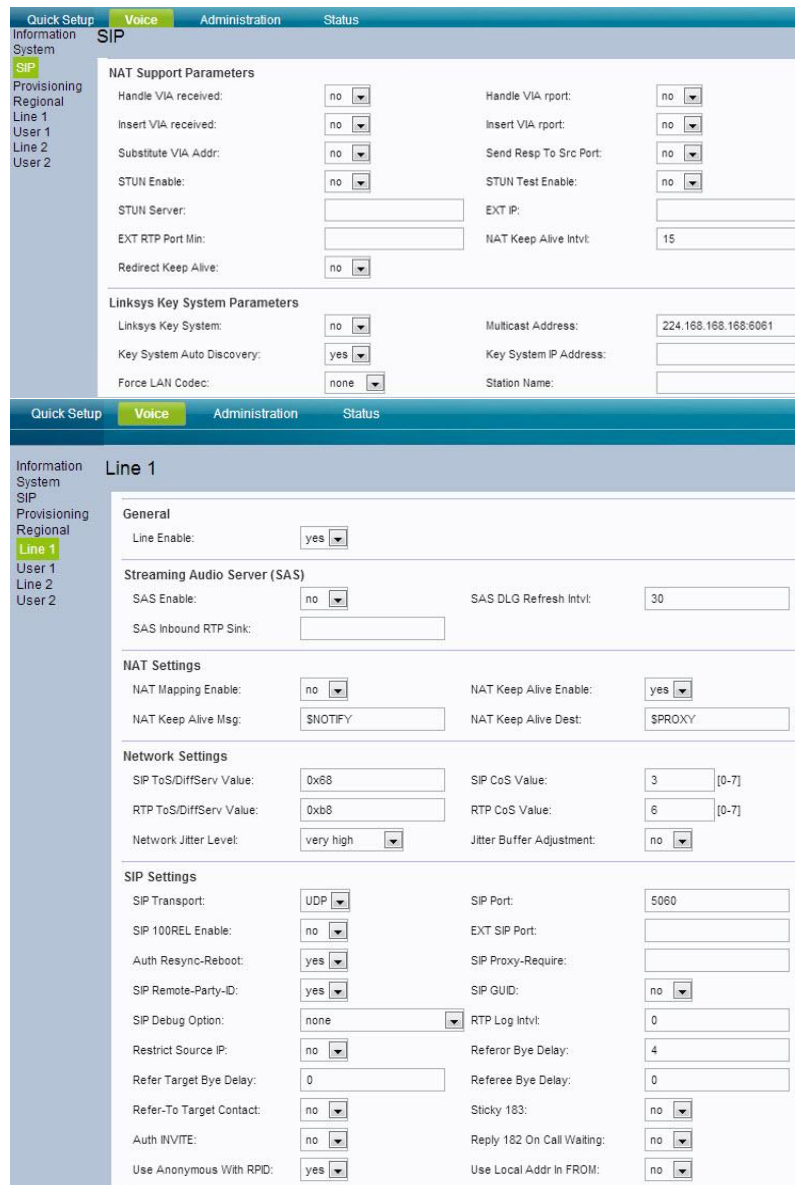
Quick Setup	Voice	Administration	Status
Information System SIP			
Response Status Code Handling			
SIT1 RSC:		SIT2 RSC:	
SIT3 RSC:		SIT4 RSC:	
Try Backup RSC:		Retry Reg RSC:	
RTP Parameters			
RTP Port Min:	16384	RTP Port Max:	16482
RTP Packet Size:	0.020	Max RTP ICMP Err:	0
RTCP Tx Interval:	0	No UDP Checksum:	no
Stats In B'VE:	yes		
SDP Payload Types			
NSE Dynamic Payload:	100	AVT Dynamic Payload:	101
INFOREQ Dynamic Payload:		G726r32 Dynamic Payload:	2
G729b Dynamic Payload:	99	EncapRTP Dynamic Payload:	112
RTP-Start-Loopback Dynamic Payload:	113	RTP-Start-Loopback Codec:	G711a
NSE Codec Name:	NSE	AVT Codec Name:	telephone-event
G711u Codec Name:	PCMU	G711a Codec Name:	PCMA
G726r32 Codec Name:	G726-32	G729a Codec Name:	G729a
G729b Codec Name:	G729ab	EncapRTP Codec Name:	encaprtp

→ **Schritt 4.3**
SIP Einstellungen Teil 3
(NAT Support Parameters, Linksys
Key System Parameters)

Passen Sie die Konfiguration unter „Voice → SIP“ gemäss der Abbildung rechts an.

→ **Schritt 5.1**
Line 1 Einstellungen Teil 1
(General, Streaming Audio Server
(SAS), NAT Settings, Network
Settings, SIP Settings)

Passen Sie die Konfiguration unter „Voice → Line 1“ gemäss der Abbildung rechts an.



The image shows two screenshots of the Cisco SPA-122 configuration web interface. The top screenshot is for the 'SIP' configuration page, and the bottom screenshot is for the 'Line 1' configuration page.

SIP Configuration:

- NAT Support Parameters:**
 - Handle VIA received: no
 - Insert VIA received: no
 - Substitute VIA Addr: no
 - STUN Enable: no
 - STUN Server: [empty]
 - EXT RTP Port Min: [empty]
 - Redirect Keep Alive: no
 - Handle VIA rport: no
 - Insert VIA rport: no
 - Send Resp To Src Port: no
 - STUN Test Enable: no
 - EXT IP: [empty]
 - NAT Keep Alive Intvl: 15
- Linksys Key System Parameters:**
 - Linksys Key System: no
 - Key System Auto Discovery: yes
 - Force LAN Codec: none
 - Multicast Address: 224.168.168.6061
 - Key System IP Address: [empty]
 - Station Name: [empty]

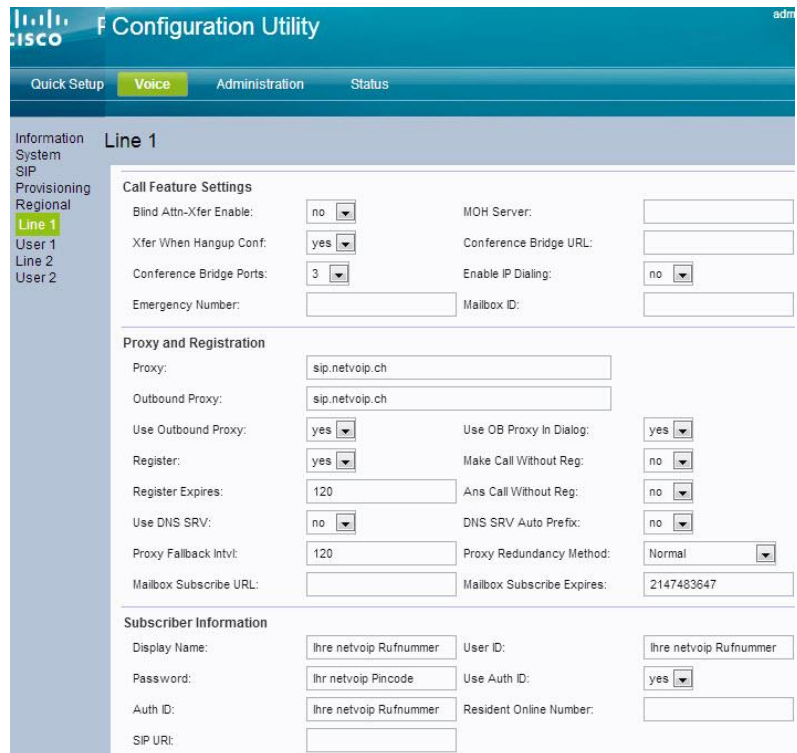
Line 1 Configuration:

- General:**
 - Line Enable: yes
- Streaming Audio Server (SAS):**
 - SAS Enable: no
 - SAS Inbound RTP Sink: [empty]
 - SAS DLG Refresh Intvl: 30
- NAT Settings:**
 - NAT Mapping Enable: no
 - NAT Keep Alive Enable: yes
 - NAT Keep Alive Msg: \$NOTIFY
 - NAT Keep Alive Dest: \$PROXY
- Network Settings:**
 - SIP ToS/DiffServ Value: 0x88
 - RTP ToS/DiffServ Value: 0xb8
 - Network Jitter Level: very high
 - SIP CoS Value: 3 [0-7]
 - RTP CoS Value: 6 [0-7]
 - Jitter Buffer Adjustment: no
- SIP Settings:**
 - SIP Transport: UDP
 - SIP 100REL Enable: no
 - Auth Resync-Reboot: yes
 - SIP Remote-Party-ID: yes
 - SIP Debug Option: none
 - Restrict Source IP: no
 - Refer Target Bye Delay: 0
 - Refer-To Target Contact: no
 - Auth INVITE: no
 - Use Anonymous With RPID: yes
 - SIP Port: 5060
 - EXT SIP Port: [empty]
 - SIP Proxy-Require: [empty]
 - SIP GUID: no
 - RTP Log Intvl: 0
 - Referor Bye Delay: 4
 - Referee Bye Delay: 0
 - Sticky 183: no
 - Reply 182 On Call Waiting: no
 - Use Local Addr In FROM: no

➔ **Schritt 5.2**
Line 1 Einstellungen Teil 2
(Call Feature Settings, Proxy and
Registration, Subscriber
Information)

Passen Sie ihre Einstellungen unter „Voice → Line 1“ gemäss der Abbildung rechts an.

Sie finden Ihre netvoip Rufnummer und Ihren netvoip Pincode in Ihrem Account auf netvoip.ch unter dem Menüpunkt „Konfiguration“ → „Geräte / Zugangsdaten“.



Line 1

Call Feature Settings

Blind Attn-Xfer Enable:	no	MOH Server:	
Xfer When Hangup Conf:	yes	Conference Bridge URL:	
Conference Bridge Ports:	3	Enable IP Dialing:	no
Emergency Number:		Mailbox ID:	

Proxy and Registration

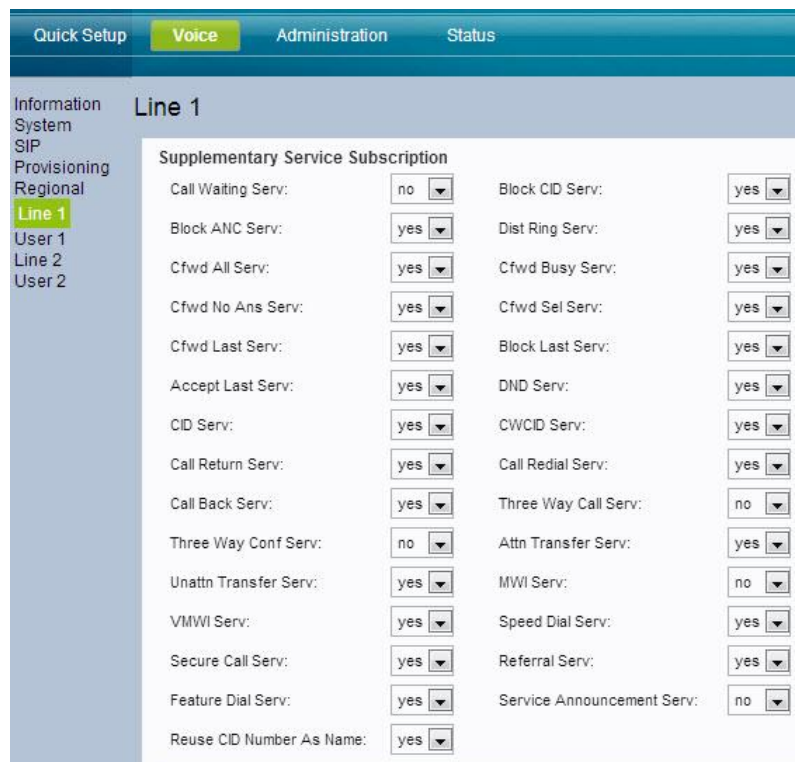
Proxy:	sip.netvoip.ch		
Outbound Proxy:	sip.netvoip.ch		
Use Outbound Proxy:	yes	Use OB Proxy in Dialog:	yes
Register:	yes	Make Call Without Reg:	no
Register Expires:	120	Ans Call Without Reg:	no
Use DNS SRV:	no	DNS SRV Auto Prefix:	no
Proxy Fallback Intvl:	120	Proxy Redundancy Method:	Normal
Mailbox Subscribe URL:		Mailbox Subscribe Expires:	2147483647

Subscriber Information

Display Name:	Ihre netvoip Rufnummer	User ID:	Ihre netvoip Rufnummer
Password:	Ihr netvoip Pincode	Use Auth ID:	yes
Auth ID:	Ihre netvoip Rufnummer	Resident Online Number:	
SIP URI:			

➔ **Schritt 5.3**
Line 1 Einstellungen Teil 3
(Supplementary Service
Subscription)

Passen Sie ihre Einstellungen unter „Voice → Line 1“ gemäss der Abbildung rechts an.



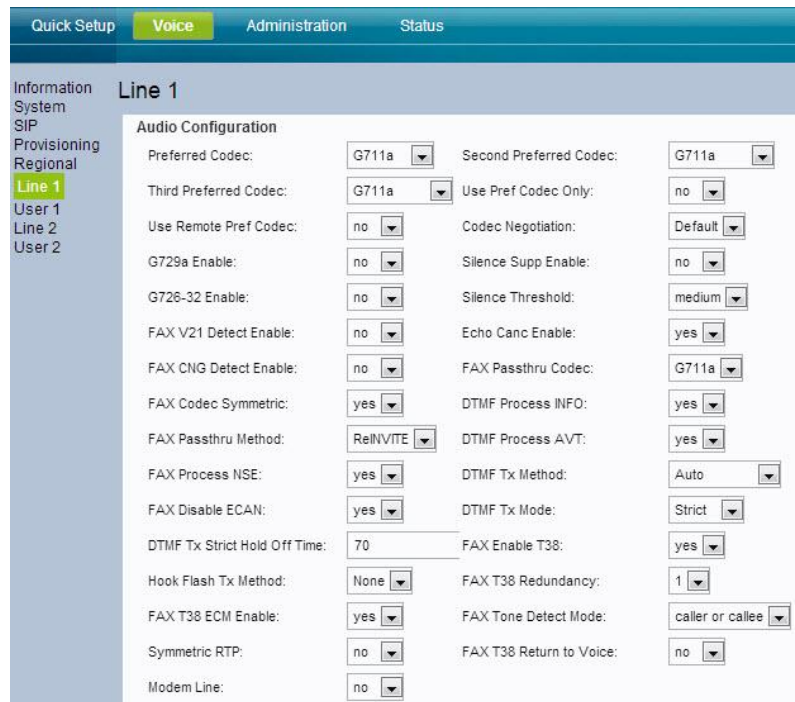
Line 1

Supplementary Service Subscription

Call Waiting Serv:	no	Block CID Serv:	yes
Block ANC Serv:	yes	Dist Ring Serv:	yes
Cfwd All Serv:	yes	Cfwd Busy Serv:	yes
Cfwd No Ans Serv:	yes	Cfwd Sel Serv:	yes
Cfwd Last Serv:	yes	Block Last Serv:	yes
Accept Last Serv:	yes	DND Serv:	yes
CID Serv:	yes	CWCID Serv:	yes
Call Return Serv:	yes	Call Redial Serv:	yes
Call Back Serv:	yes	Three Way Call Serv:	no
Three Way Conf Serv:	no	Attn Transfer Serv:	yes
Unattn Transfer Serv:	yes	MWI Serv:	no
VMWI Serv:	yes	Speed Dial Serv:	yes
Secure Call Serv:	yes	Referral Serv:	yes
Feature Dial Serv:	yes	Service Announcement Serv:	no
Reuse CID Number As Name:	yes		

➔ **Schritt 5.4**
Line 1 Einstellungen Teil 4
(Audio Configuration)

Passen Sie die Konfiguration unter „Voice → Line 1“ gemäss der Abbildung rechts an.

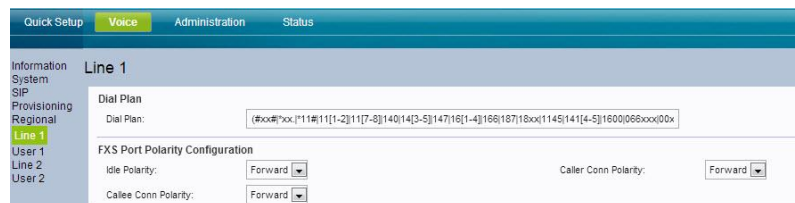


The screenshot shows the 'Voice' configuration page for 'Line 1'. The 'Audio Configuration' section includes the following settings:

- Preferred Codec: G711a
- Second Preferred Codec: G711a
- Third Preferred Codec: G711a
- Use Remote Pref Codec: no
- Use Pref Codec Only: no
- G729a Enable: no
- Codec Negotiation: Default
- G726-32 Enable: no
- Silence Supp Enable: no
- Silence Threshold: medium
- FAX V21 Detect Enable: no
- Echo Canc Enable: yes
- FAX CNG Detect Enable: no
- FAX Passthru Codec: G711a
- FAX Codec Symmetric: yes
- DTMF Process INFO: yes
- FAX Passthru Method: ReINVITE
- DTMF Process AVT: yes
- FAX Process NSE: yes
- DTMF Tx Method: Auto
- FAX Disable ECAN: yes
- DTMF Tx Mode: Strict
- DTMF Tx Strict Hold Off Time: 70
- FAX Enable T38: yes
- Hook Flash Tx Method: None
- FAX T38 Redundancy: 1
- FAX T38 ECM Enable: yes
- FAX Tone Detect Mode: caller or callee
- Symmetric RTP: no
- FAX T38 Return to Voice: no
- Modem Line: no

➔ **Schritt 5.5**
Line 1 Einstellungen Teil 5
(Dial Plan)

Passen Sie den Dial Plan unter „Voice → Line 1“ gemäss der Abbildung rechts an. Den kompletten Dialplan entnehmen Sie bitte der unten aufgeführten Kopiervorlage:



The screenshot shows the 'Dial Plan' configuration for 'Line 1'. The 'Dial Plan' field contains the following text:

```
(#xx#|*xx.|*11#[11[1-2]11[7-8]14014[3-5]14716[1-4]16618718xx|1145141[4-5]1600066xxx|00xxxxxx.|01xxxxxx|02[1-2]xxxxxx|024xxxxxx|02[6-7]xxxxxx|03[1-4]xxxxxx|041xxxxxx|04[3-4]xxxxxx|05[1-2]xxxxxx|05[5-6]xxxxxx|058xxxxxx|06[1-2]xxxxxx|071xxxxxx|074xxxxxx|075xxxxxx|07[6-9]xxxxxx|081xxxxxx|084xxxxxx|091xxxxxx|0800xxxxxx|0878xxxxxx|090[0-1]xxxxxx|0906xxxxxx|0860xxxxxx|10xxxxxxxxxxxxxx.)
```

The 'FXS Port Polarity Configuration' section shows:

- Idle Polarity: Forward
- Caller Conn Polarity: Forward
- Callee Conn Polarity: Forward

```
(#xx#|*xx.|*11#[11[1-2]11[7-8]14014[3-5]14716[1-4]16618718xx|1145141[4-5]1600066xxx|00xxxxxx.|01xxxxxx|02[1-2]xxxxxx|024xxxxxx|02[6-7]xxxxxx|03[1-4]xxxxxx|041xxxxxx|04[3-4]xxxxxx|05[1-2]xxxxxx|05[5-6]xxxxxx|058xxxxxx|06[1-2]xxxxxx|071xxxxxx|074xxxxxx|075xxxxxx|07[6-9]xxxxxx|081xxxxxx|084xxxxxx|091xxxxxx|0800xxxxxx|0878xxxxxx|090[0-1]xxxxxx|0906xxxxxx|0860xxxxxx|10xxxxxxxxxxxxxx.)
```

→ Schritt 6 (optional) Fax Einstellungen

Falls Sie Ihre netvoip Nummer als Faxnummer einrichten möchten, ändern Sie bitte folgende Parameter gemäss Abbildung rechts.

Information		Line 1	
System	SIP	Audio Configuration	
Provisioning	Regional	Preferred Codec:	G711a
Line 1		Third Preferred Codec:	Unspecified
User 1		Use Remote Pref Codec:	no
User 2		G729a Enable:	no
		G726-32 Enable:	no
		FAX V21 Detect Enable:	yes
		FAX CNG Detect Enable:	yes
		FAX Codec Symmetric:	yes
		FAX Passthru Method:	RelINVITE
		FAX Process NSE:	yes
		FAX Disable ECAN:	yes
		DTMF Tx Strict Hold Off Time:	70
		Hook Flash Tx Method:	None
		FAX T38 ECM Enable:	yes
		Symmetric RTP:	no
		Second Preferred Codec:	Unspecified
		Use Pref Codec Only:	no
		Codec Negotiation:	Default
		Silence Supp Enable:	no
		Silence Threshold:	medium
		Echo Canc Enable:	yes
		FAX Passthru Codec:	G711a
		DTMF Process INFO:	yes
		DTMF Process AVT:	yes
		DTMF Tx Method:	Auto
		DTMF Tx Mode:	Strict
		FAX Enable T38:	yes
		FAX T38 Redundancy:	1
		FAX Tone Detect Mode:	caller or callee
		FAX T38 Return to Voice:	no

Installation beendet

Gratulation.
Sie haben die Installation abgeschlossen. Wir wünschen Ihnen viel Spass mit dem neuen Cisco SPA-122.

Technischer Support Kontakt

Benötigen Sie Hilfe bei der Installation? Zögern Sie nicht uns zu kontaktieren. Sie erreichen unseren technischen Support von Montag - Freitag zwischen 08:00 Uhr und 18:00 Uhr unter der Nummer 0800 700 627 oder per E-Mail unter support@netstream.ch.

Besten Dank.