

February 18, 2020

Configure MiVoice Office 400 for use with Backbone Solutions AG sipcall SIP trunks

Description: This document describes how to configure MiVoice Office 400 with sipcall by Backbone Solution AG SIP Trunks.

Environment: Mitel MiVoice 400 Release 6.1 SP2, Porta MR60 als SIP/Media-Gateway MR60, Oracle AP4600 Session Boarder Controller SCZ8.1.0

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Mitel Technical Configuration Notes:

Configure MiVoice Office 400 to use with sipcall Backbone Solutions AG sipcall SIP trunks

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Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Office 400 to connect to the sipcall by Backbone Solution AG Sip trunk. Different components can be configured in various configurations depending on your VoIP solution. This document covers a basic setup with basic SIP trunk routing rules.

Interop History

Version	Date	Reason
1	November, 2019	Initial Interop with Mitel MiVoice Office 400 Release 6.1 and sipcall by Backbone Solution AG SIP trunk.

Interop Status

This Interop of sipcall by Backbone Solution AG SIP trunk with MiVoice Office 400 has been given a Compatible Certification status. This SIP trunk will be included in the SIP CoE Reference Guide.

 COMPATIBLE	The most common certification which means MiVoice Office 400 has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.
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Software & Hardware Setup

The table below provides the hardware and software specifications used to generate SIP Audio calls, both point to point and conference calls, using sipcall by Backbone Solution AG SIP trunk connected to MiVoice Office 400.

Manufacturer	Variant	Software Version
Mitel	MiVoice Office 470	Release 6.1 SP2 (8971a1)
Mitel	68xx series SIP sets	5.1.0.2047
Sipcall by Backbone Solutions AG	Porta MR60 als SIP/Media-Gateway	MR60
Sipcall by Backbone Solutions AG	Oracle AP4600 Session Boarder Controller.	SCZ8.1.0

Tested Features

The below table provides an overview of the features tested during the Interoperability test cycle and is not a detailed view of the test cases. Please see the SIP Trunk Side Interoperability Test Plan for detailed test cases and results.

Feature	Feature Description	Issues
Basic Call	Making and Receiving a call through sipcall by Backbone Solution AG for all conferencing scenarios, long calls durations, call hold, call transfer, Call Forward, Call Park	<input checked="" type="checkbox"/>
DTMF	Test cases were performed using OOB DTMF	<input checked="" type="checkbox"/>
Caller ID	Blocking and Unblocking Caller ID scenarios tested	<input checked="" type="checkbox"/>
Fax	Fax calls using G.711 codec.	<input type="checkbox"/>
Fax	T.38 Fax calls.	<input type="checkbox"/>
Video	Making and receiving a call through MiVoice Office 400 with video capable devices.	<input type="checkbox"/>

- No issues found - Issues found, cannot recommend to use - Issues found Not tested

Network Topology

This diagram shows how the testing network is configured for reference.

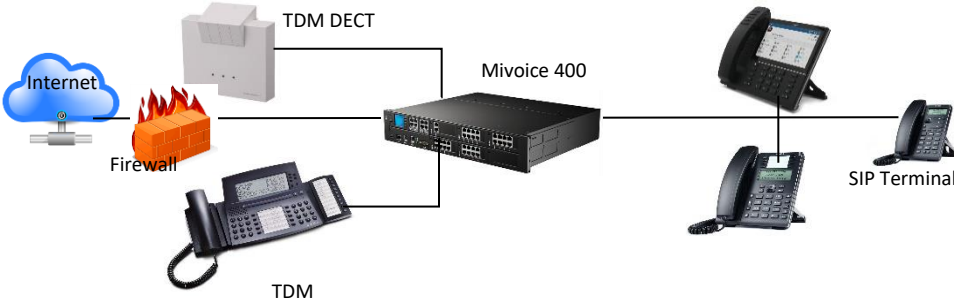


Figure 1 – Network Topology

Device Limitations and Known Issues

This is a list of not tested or not supported features when SIP trunk is connected to the MiVoice 400.

Feature	Problem Description
G729 Codec	Not Tested Recommendation: Contact sipcall for more information
SIP-INFO, In-band DTMF	Not Tested Recommendation: Contact sipcall for more information
Fax	Not Tested Recommendation: Contact sipcall for more information
TLS/SRTP	sipcall does not support TLS/SRTP Recommendation: Contact sipcall for more information
Resiliency	MiVO 400 doesn't support 3 rd party resiliency. Recommendations: Contact Mitel for more details.
Video calling	Not Tested vRecommendation: Contact sipcall for more information

Configuration Notes

This section is a description of how the SIP Interop network was configured. These notes provide a guideline as how a device can be configured in a customer environment and how the sipcall and MiVO 400 were configured in our test environment.

Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.

MiVoice Office 400 Configuration Notes

The following information shows how to configure a MiVoice Office 400 to interconnect with sipcall by Backbone Solution AG SIP trunk.

Network Requirements

- There must be adequate bandwidth to support the VoIP network. As a guide, the Ethernet bandwidth is approx 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

Assumptions for MiVoice Office 400 Programming

- The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that MiVoice Office 400 is equipped with enough SIP Access Channel licenses for the connection to sipcall by Backbone Solution AG. Up to 30 SIP voice channels are available for each SIP provider. For each SIP voice channel, you need a SIP Access Channels license.

Network Interfaces

Create a network interface for a SIP trunk provider, c. In this example, sipcall by Backbone Solution AG is reachable using an IP address as entered in the “Registrar IP address” field. Your configuration may be different depending on the type and configuration of the SBC you are using. In our test configuration.

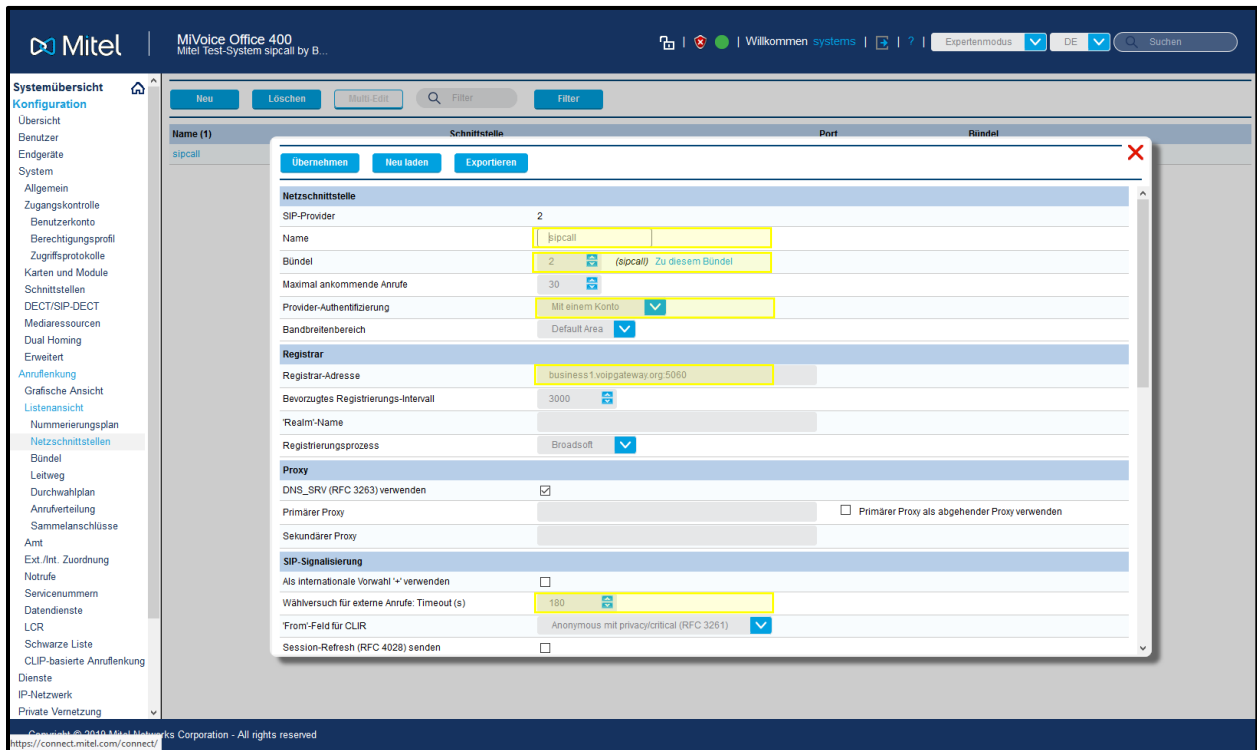


Figure 2 – Network Interface Creation

Network Interface Settings

The following 2 figures show the settings that were used for establishing a connection to sipcall by Backbone Solutions AG SIP trunk. Most of the settings were left at their default values. You may want to specify a preferred codec. does not support PRACK so disable this. The final thing to do is to create SIP account. You can use one of the DN's supplied by sipcall by Backbone Solutions AG for this purpose.

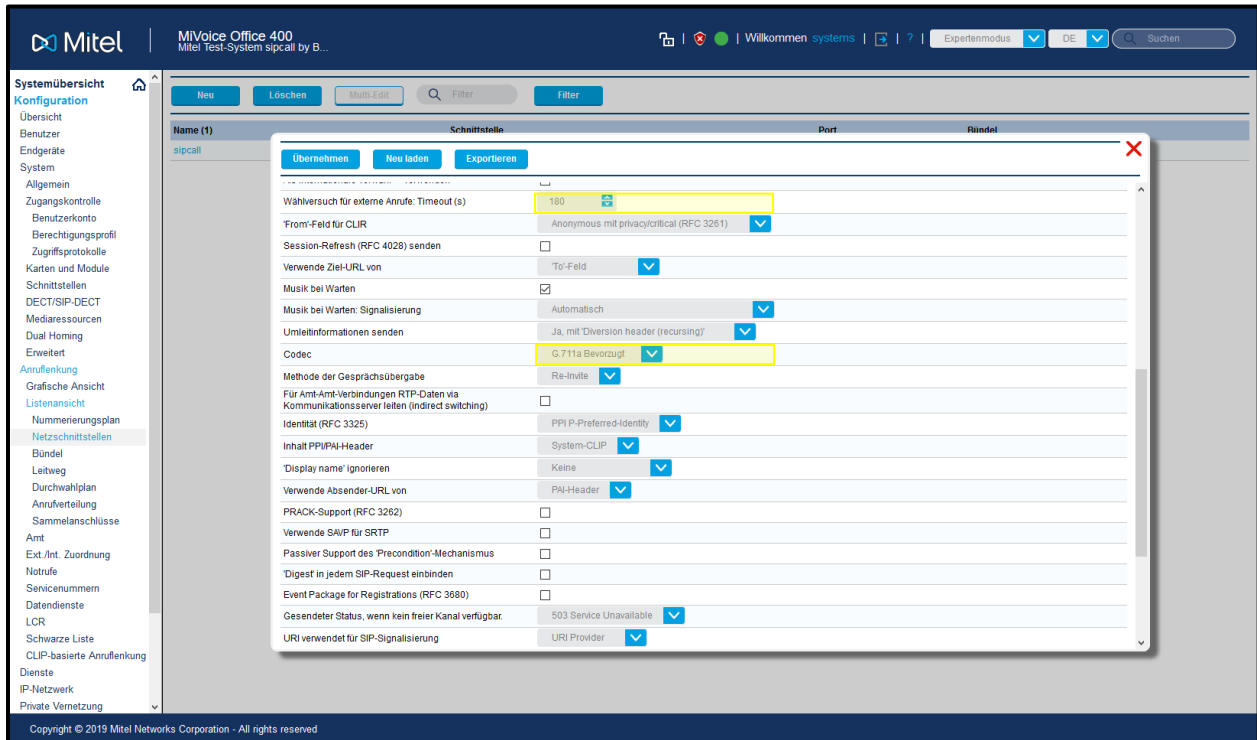


Figure 3 – Network Interface Settings

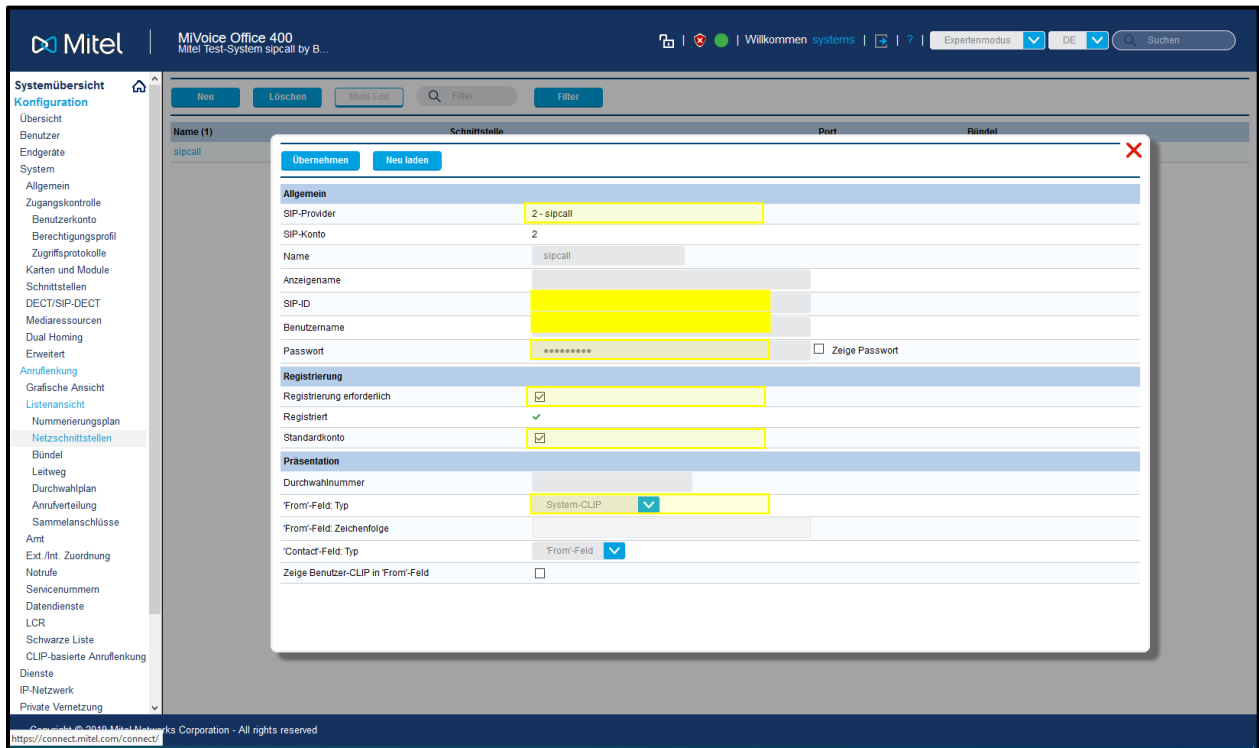
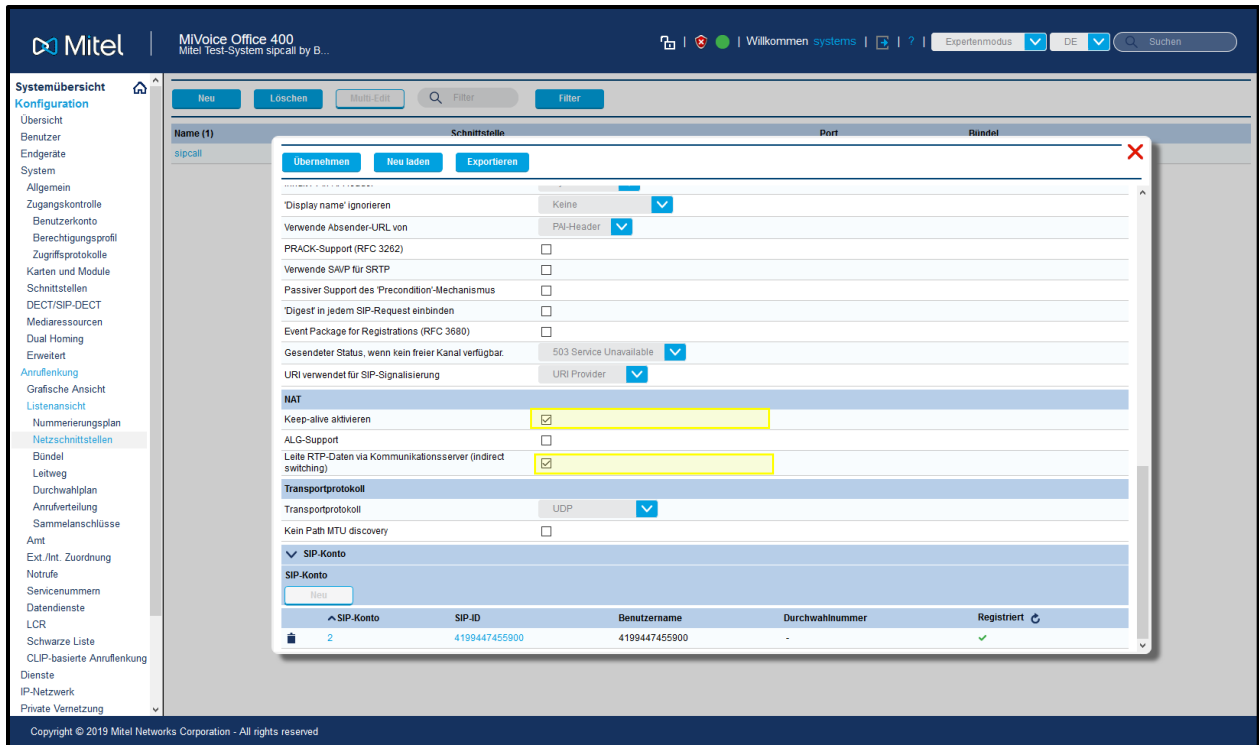


Figure 4 – Network Interface Setting (Continued)

Outgoing Call Routing

Create a route to handle your outgoing calls. In the test setup route 1 was used for outgoing calls to c with a call number of 9. This will route all calls that begin with the digit 9 to the sipcall by Backbone Solutions AG interface, **See figure 5.**

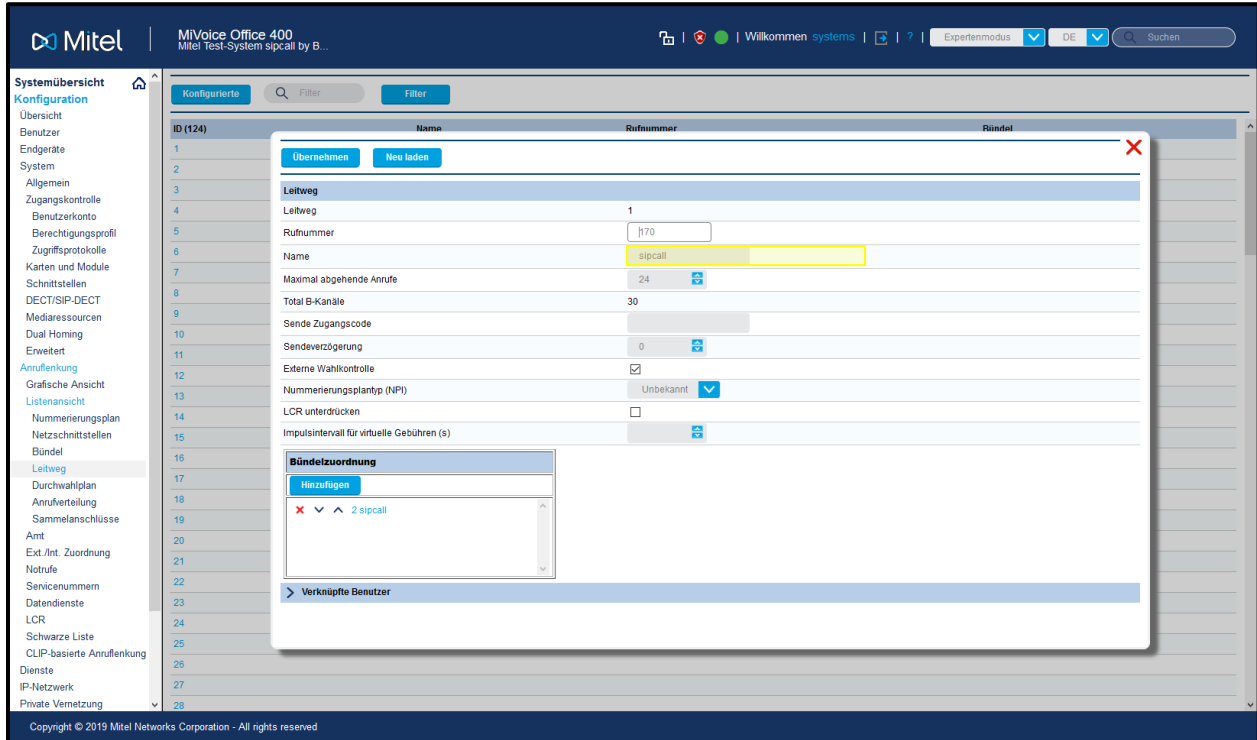


Figure 5 – Trunk Service Assignment

Incoming Call Route

There are several different ways to route inbound calls to a destination answer point. Inbound calls were tested using a DDI plan and a Call Distribution Element. As well, calls were routed to both a User Group and individual users. Calls can be directed to a DDI plan or a CDE via the Trunk Group we created when the Network interface was created.

The screenshot displays the Mitel MiVoice Office 400 configuration interface. The left sidebar shows the 'Systemübersicht' (System Overview) menu with various configuration categories. The main content area shows the configuration for a network interface (ID 1) with a 'Vernetzung' (Networking) section. The 'Anrufkennung (CLIP)' (Call Identification) section is expanded, showing the following settings:

Section	Setting	Value
Vernetzung	Netztyp	Öffentlich
	Ruf wenn NPI 'Unknown'	Extern
	CLIP abschneiden	
	NPI überschreiben	Nein
Rufkontrollen	Rufkontrollen für ankommende Anrufe	Nicht generieren
	Rufkontrollen für abgehende Anrufe	Generieren
Anrufkennung (CLIP)	Sofortige Auslösung bei besetzt senden	<input type="checkbox"/>
	Called Party Number Format	Kürzestmöglich
Leistungsmerkmale	Benachrichtigung	
	Notifikationen senden	<input checked="" type="checkbox"/>
	Umleitinformationen senden	<input checked="" type="checkbox"/>
Integration mobiles/externes Telefon	ECT-Information	<input checked="" type="checkbox"/>
	CLIP-Authentifizierung auch bei ungeprüfter CLIP erlauben	<input type="checkbox"/>
Schwarze Liste	Erweiterte Funktionalität erlauben bei direkt ankommenden Anrufen	<input type="checkbox"/>
	Schwarze Liste für ankommende Anrufe anwenden	Zur Schwarzen Liste

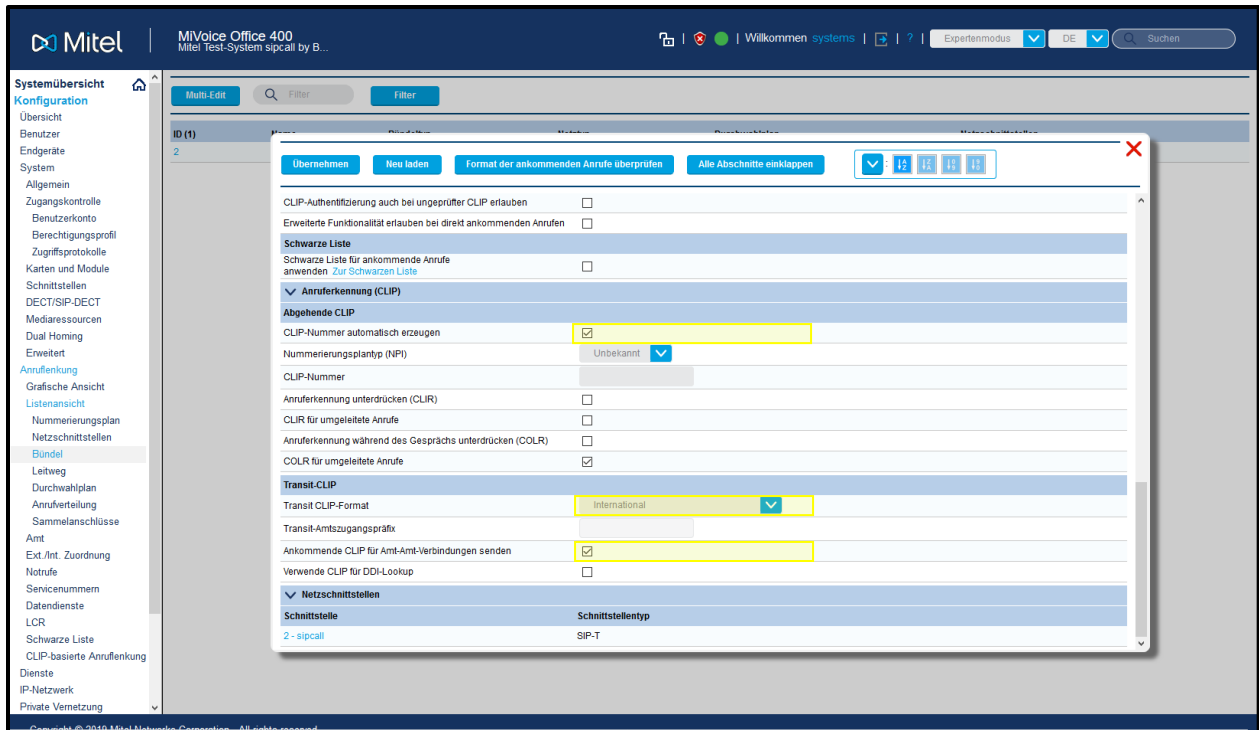


Figure 6: Trunk Group

DDI Plan

The DDI Plan is where you can assign individual called numbers to specific Users or User Groups etc. In the example below **Figure 7** two incoming numbers were created to route calls to individual destinations. These called numbers were then routed to destinations using the Call Distribution Elements as depicted in **Figure 8** below.

Durchwahlplan (18/18)	Durchwahlnummer	Anruf
1	41	101
1	41	102
1	41	103
1	41	104
1	41	105
1	41	106
1	41	107
1	41	108
1	41	100
2	41	100
2	41	101
2	41	102
2	41	103
2	41	104
2	41	105
2	41	106
2	41	107
2	41	108

Figure 7: DDI Creation

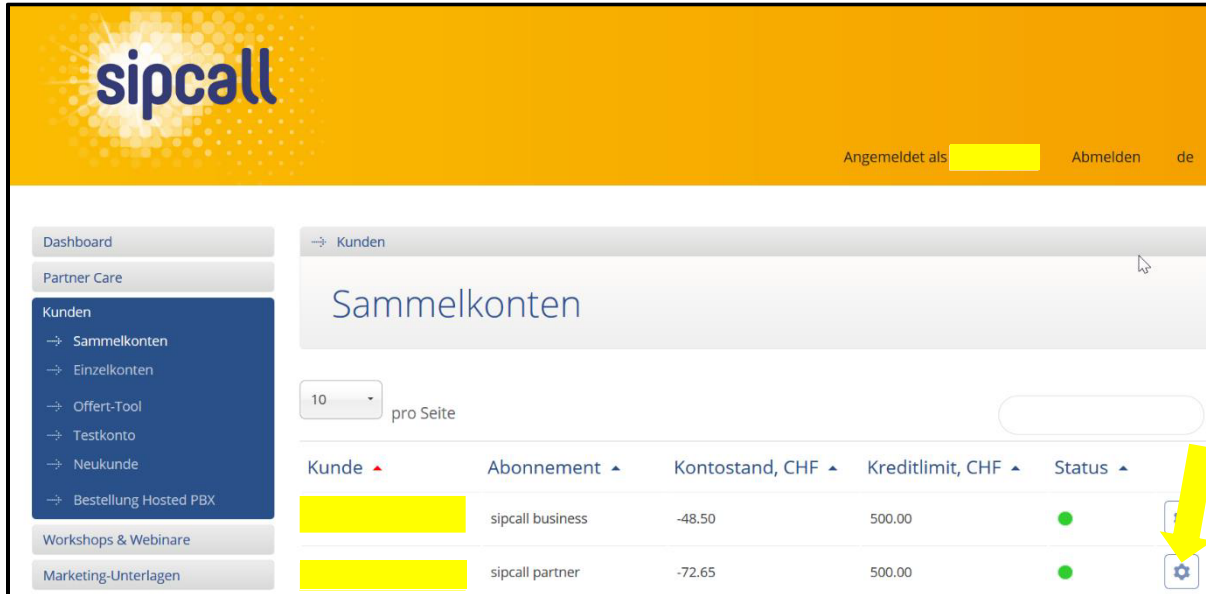
ID (20)	Name	Rufnummer	Schaltgruppe	Aktiv	Schaltposition 1	Schaltposition 2	Schaltposition 3	CFNR	CFB	Verknüpfte Durchwahlnummern
1			1	✓	• Sammelanschluss 16	Benutzer 21	Sammelanschluss 16	-	-	
100	Praxis		1	✓	• Benutzer 21 • Sammelanschluss 16	Benutzer 21	Benutzer 21 • Sammelanschluss 16	-	-	
101	Sprechzimmer 1		1	✓	• Benutzer 31	Benutzer 21	Benutzer 31	-	-	
102	Sprechzimmer 2		1	✓	• Benutzer 32	Benutzer 21	Benutzer 32	-	-	
103	Sprechzimmer 3		1	✓	• Benutzer 33	Benutzer 21	Benutzer 33	-	-	
104			1	✓	• Sammelanschluss 16	Benutzer 21	Sammelanschluss 16	-	-	
105	Sabrina Beerli		1	✓	• Benutzer 34	Benutzer 34	Benutzer 34	-	-	
106			1	✓	• Sammelanschluss 16	Benutzer 21	Sammelanschluss 16	-	-	
107			1	✓	• Sammelanschluss 16	Benutzer 21	Sammelanschluss 16	-	-	
108	Nofall		1	✓	• Sammelanschluss 1	Sammelanschluss 1	Sammelanschluss 1	-	-	
109			1	✓	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
110			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
111			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
112			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
113			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
114			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
115			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
116			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
117			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	
118			1	✗	• Sammelanschluss 16	Sammelanschluss 16	Sammelanschluss 16	-	-	

Figure 8: CDE

Provider sipcall

Below are the few screenshots for

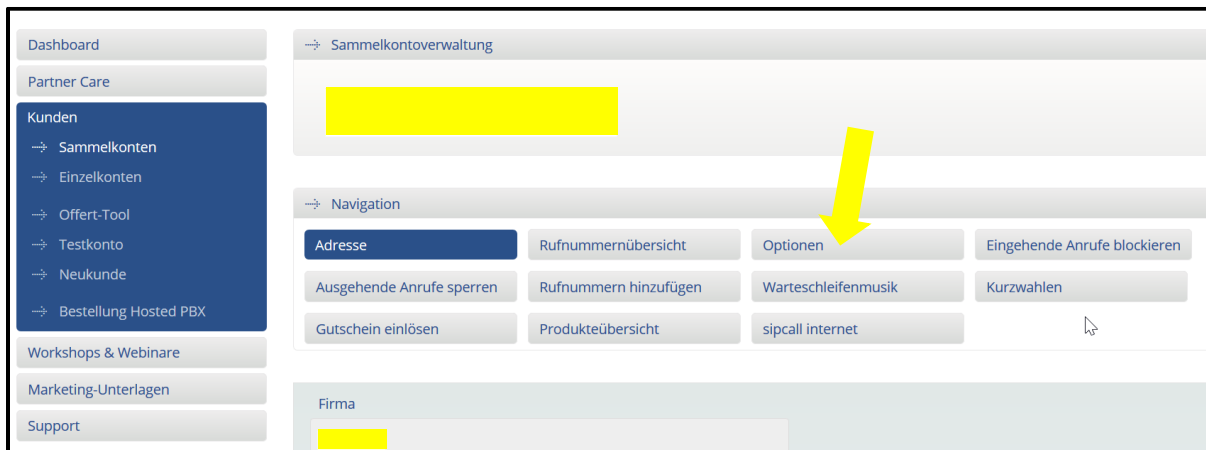
Customer selects



The screenshot shows the sipcall web interface. The top navigation bar includes the sipcall logo and the text "Angemeldet als [redacted] Abmelden de". The left sidebar contains a menu with "Kunden" selected, showing sub-items: "Sammelkonten", "Einzelkonten", "Offert-Tool", "Testkonto", "Neukunde", and "Bestellung Hosted PBX". The main content area is titled "Kunden" and "Sammelkonten". It features a table with columns: "Kunde", "Abonnement", "Kontostand, CHF", "Kreditlimit, CHF", and "Status". A yellow arrow points to a gear icon in the "Status" column of the second row.

Kunde	Abonnement	Kontostand, CHF	Kreditlimit, CHF	Status
[redacted]	sipcall business	-48.50	500.00	●
[redacted]	sipcall partner	-72.65	500.00	●

Option Trunk



The screenshot shows the sipcall web interface for account management. The top navigation bar includes the sipcall logo and the text "Angemeldet als [redacted] Abmelden de". The left sidebar contains a menu with "Kunden" selected, showing sub-items: "Sammelkonten", "Einzelkonten", "Offert-Tool", "Testkonto", "Neukunde", and "Bestellung Hosted PBX". The main content area is titled "Sammelkontoverwaltung" and shows a redacted account name. Below this is a "Navigation" section with buttons: "Adresse", "Rufnummernübersicht", "Optionen", "Eingehende Anrufe blockieren", "Ausgehende Anrufe sperren", "Rufnummern hinzufügen", "Warteschleifenmusik", "Kurzahlen", "Gutschein einlösen", "Produkteübersicht", and "sipcall internet". A yellow arrow points to the "Optionen" button. At the bottom, there is a "Firma" field with a redacted name.

Configurations detail for Input in the MiVoice 400

Konfigurationsdaten

Benutzerkennung: 41-1157913

VoIP Passwort: w28

[Neues VoIP-Passwort generieren](#)

Registrierte Endgeräte (Aktualisieren)

Endgerät	Kontakt	Registriert am (UTC)	Registriert bis (UTC)
Aastra 400	212.203.91.174:11795	2019-12-11 19:59:20	2019-12-11 20:49:20