Mitel Technical Configuration Notes – HO3541

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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# Table of Contents

# 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.

| | |
|---|---|
|   | 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. |

## 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 | ☑ |
| **DTMF** | Test cases were performed using OOB DTMF | ☑ |
| **Caller ID** | Blocking and Unblocking Caller ID scenarios tested | ☑ |
| **Fax** | Fax calls using G.711 codec. | ⊘ |
| **Fax** | T.38 Fax calls. | ⊘ |
| **Video** | Making and receiving a call through MiVoice Office 400 with video capable devices. | ⊘ |

☑ *- 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.

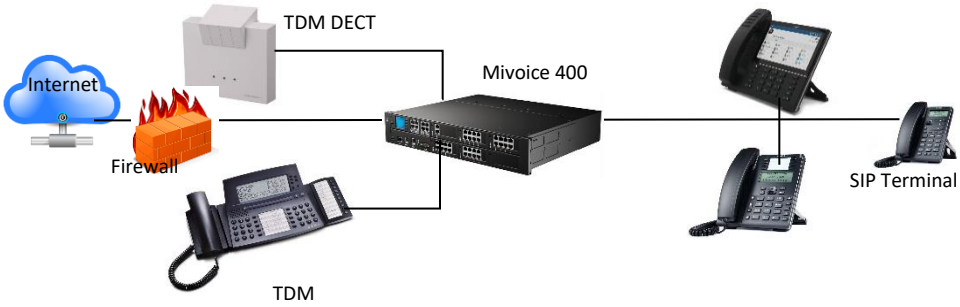



*Figure 1 – Network Topology*

### 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<br>**Recommendation:** Contact sipcall for more information |
| **SIP-INFO, In-band DTMF** | Not Tested<br>**Recommendation:** Contact sipcall for more information |
| **Fax** | Not Tested<br>**Recommendation:** Contact sipcall for more information |
| **TLS/SRTP** | sipcall does not support TLS/SRTP<br>**Recommendation:** Contact sipcall for more information |
| **Resillency** | MiVO 400 doesn't support 3rd party resiliency.<br>**Recommendations:** Contact Mitel for more details. |
| **Video calling** | Not Tested<br>**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.

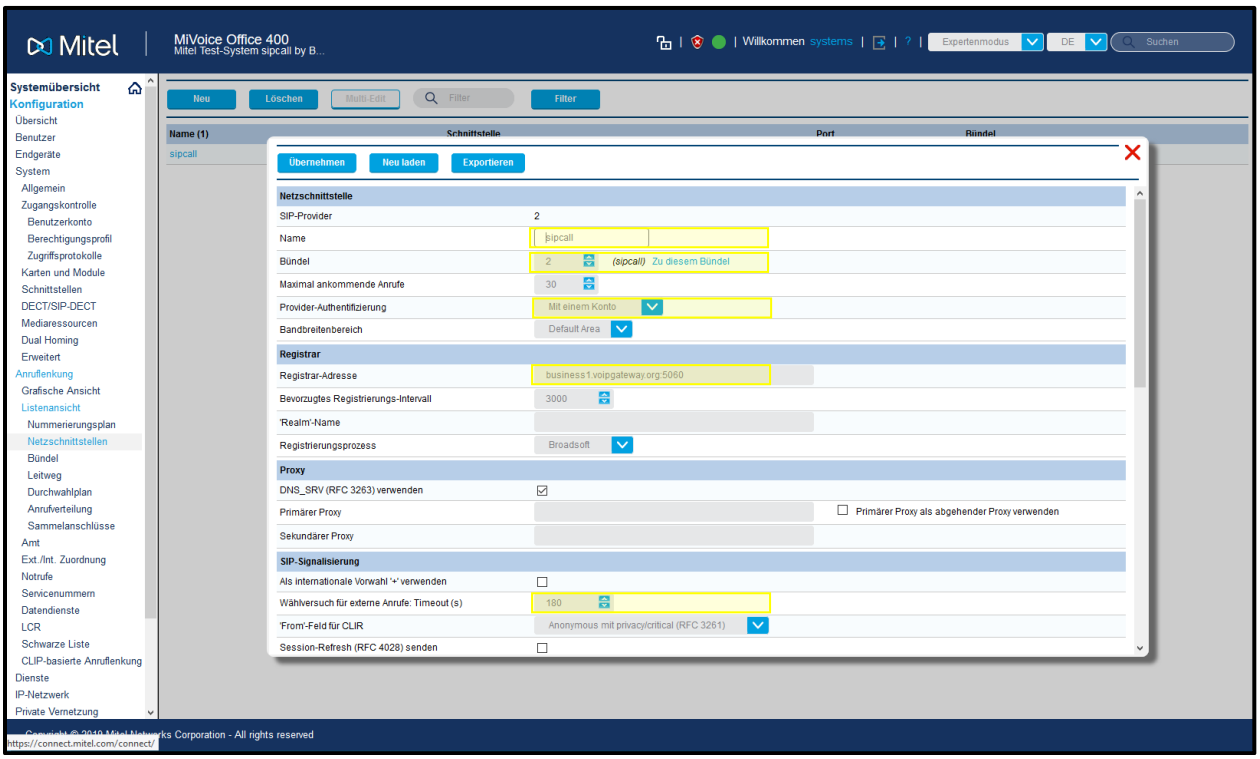



*Figure 2 – Network Interface Creation*

### *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.

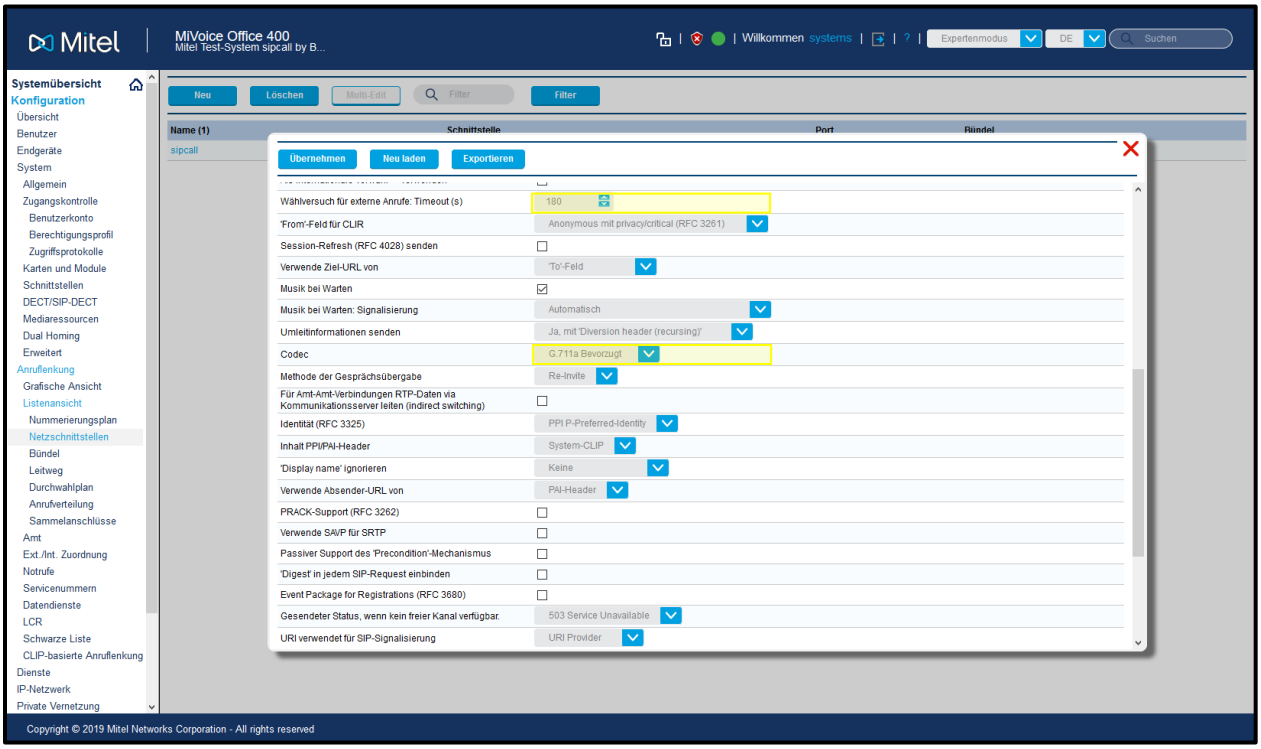
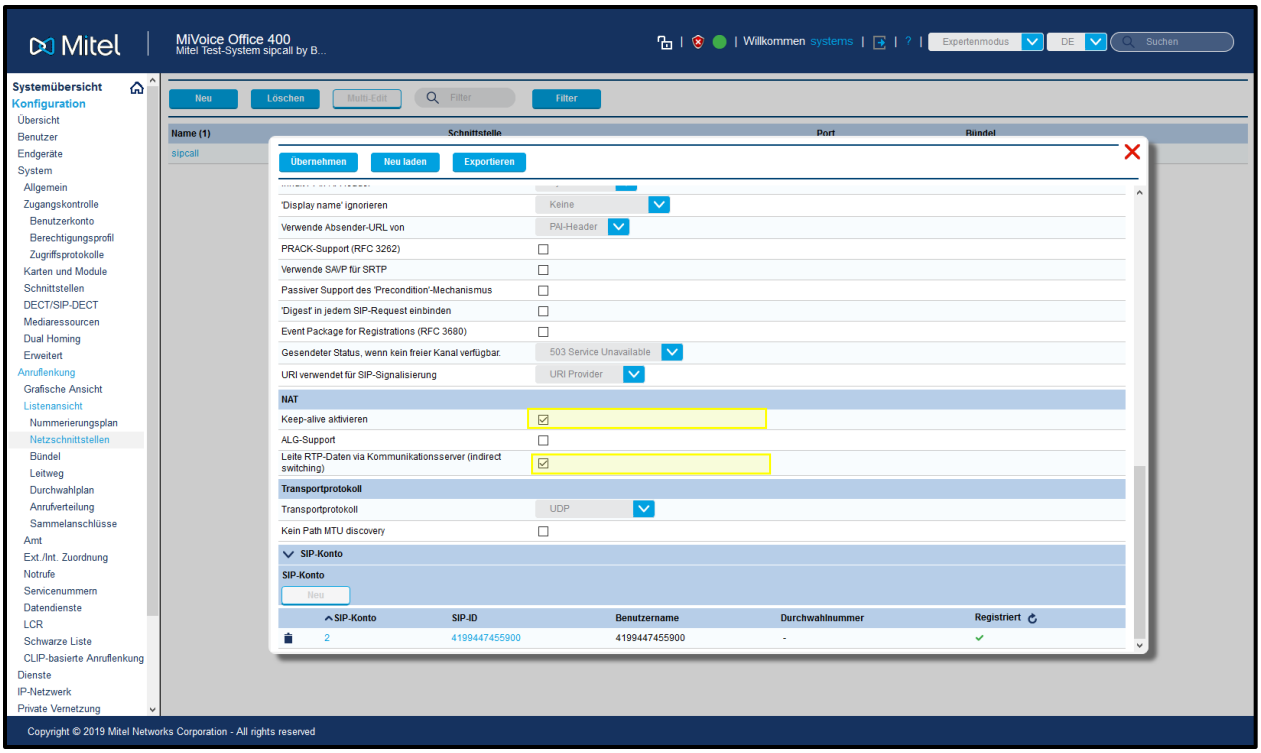



*Figure 3 – Network Interface Settings*



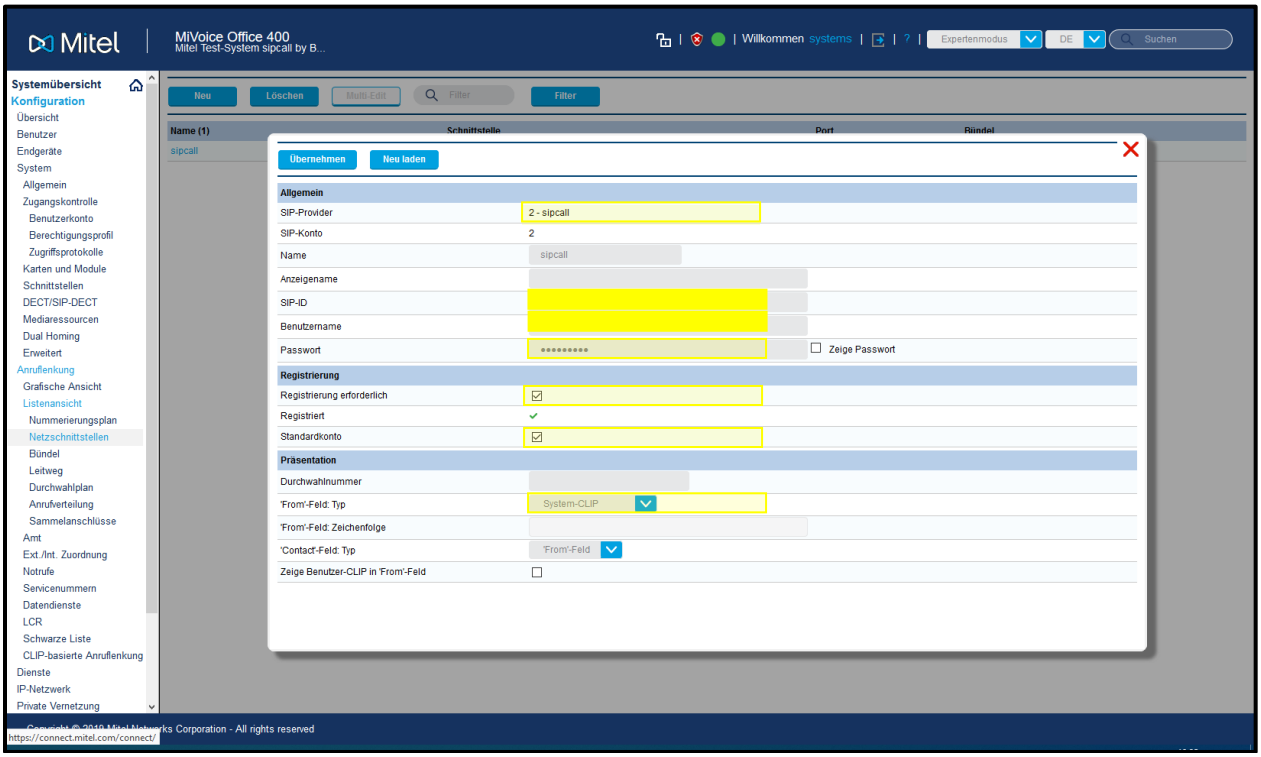




*Figure 4 – Network Interface Setting (Continued)*

### *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.**

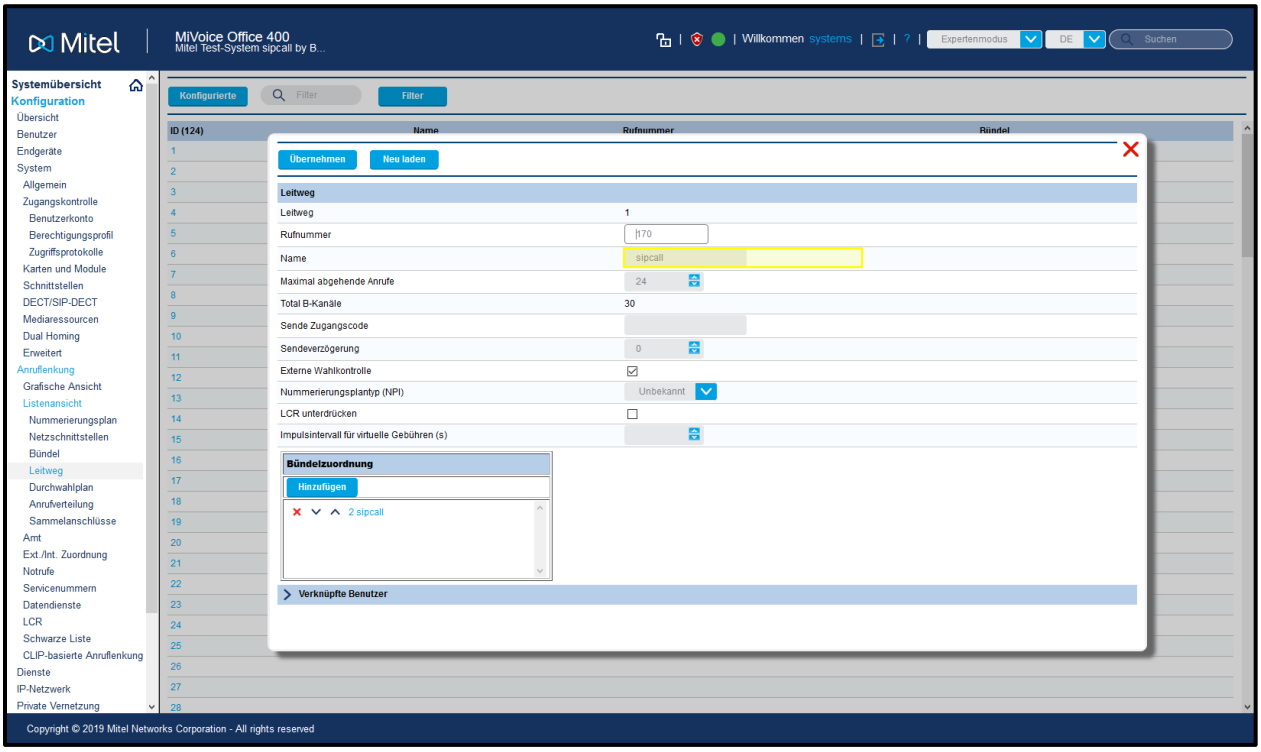
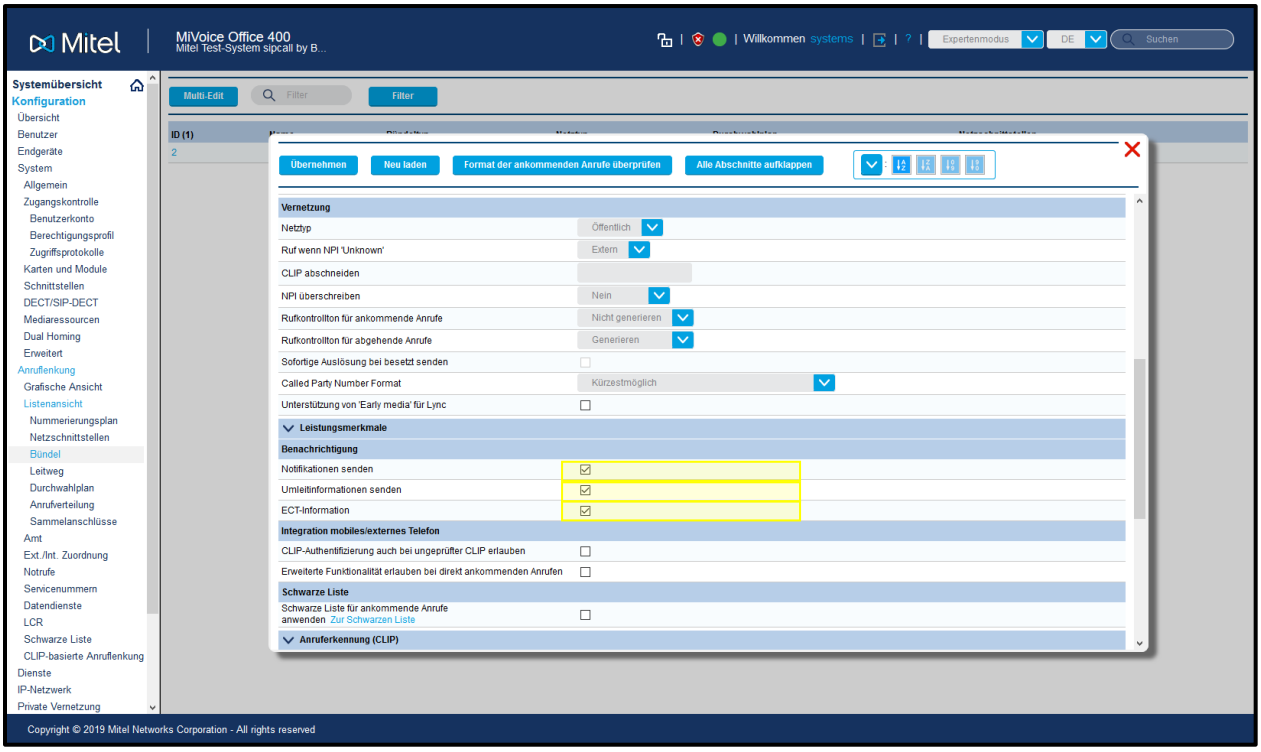



*Figure 5 – Trunk Service Assignment*

### *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.



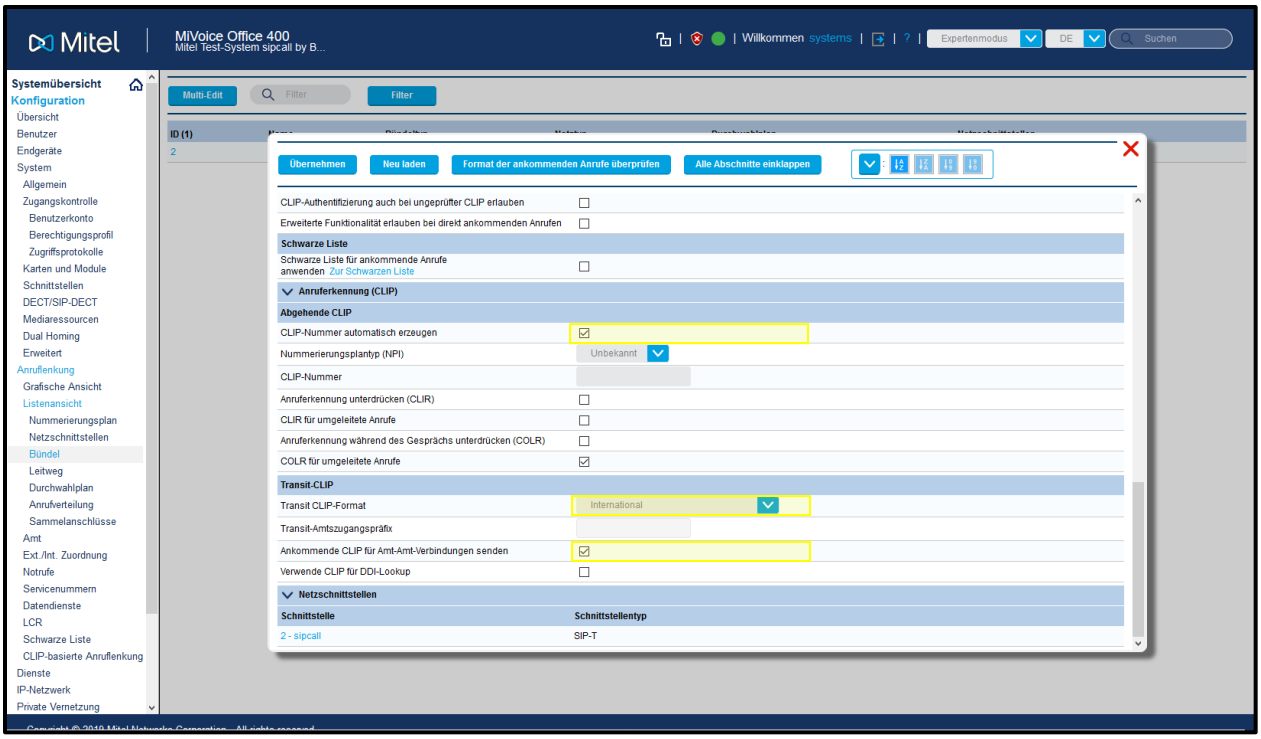



*Figure 6: Trunk Group*

### *DDI Plan*

The DDI Plan is where you can assign individual called numbers to specific Users or User Groups etc. In the example bellow **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.

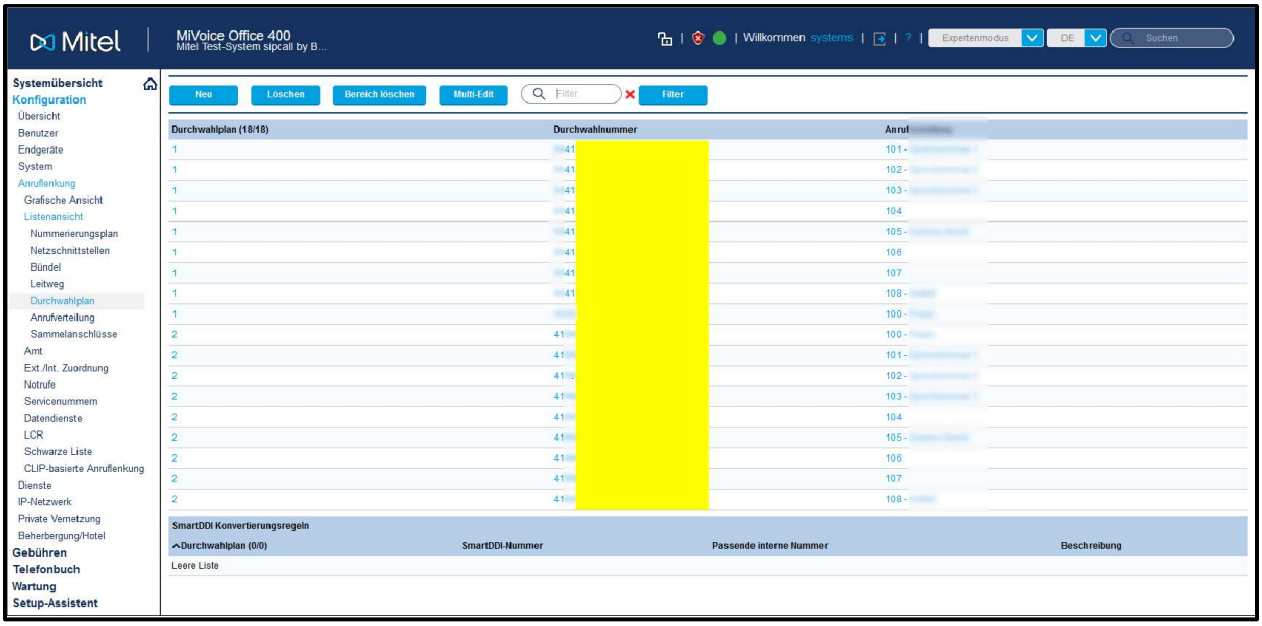



*Figure 7: DDI Creation*

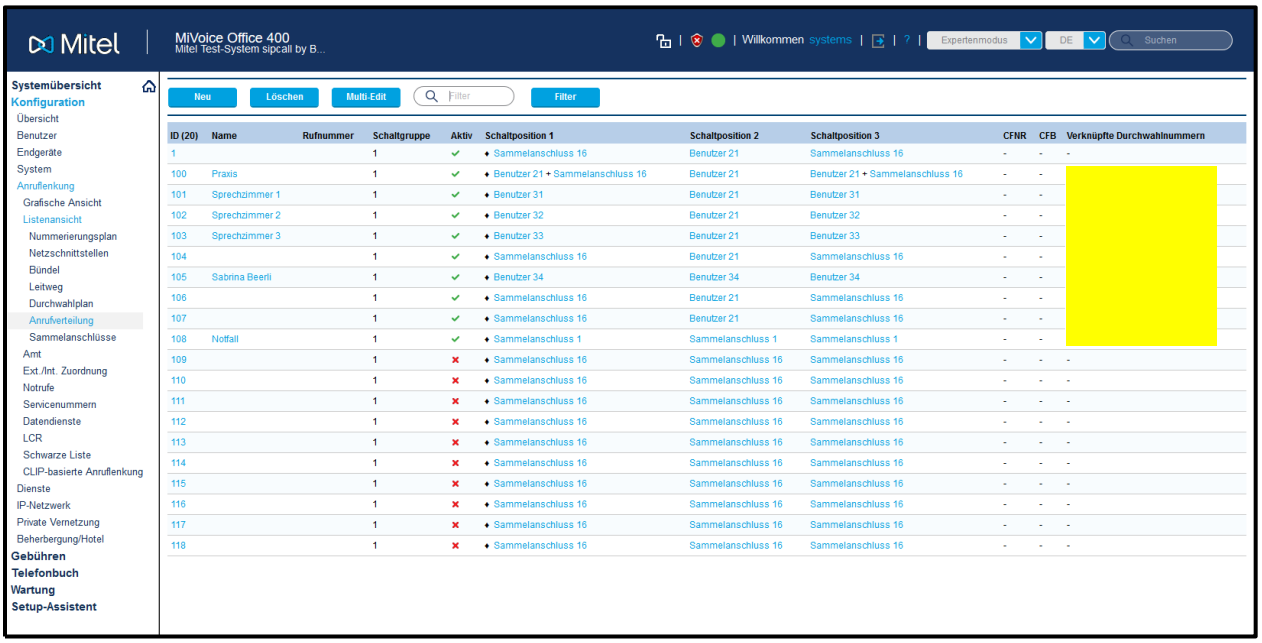
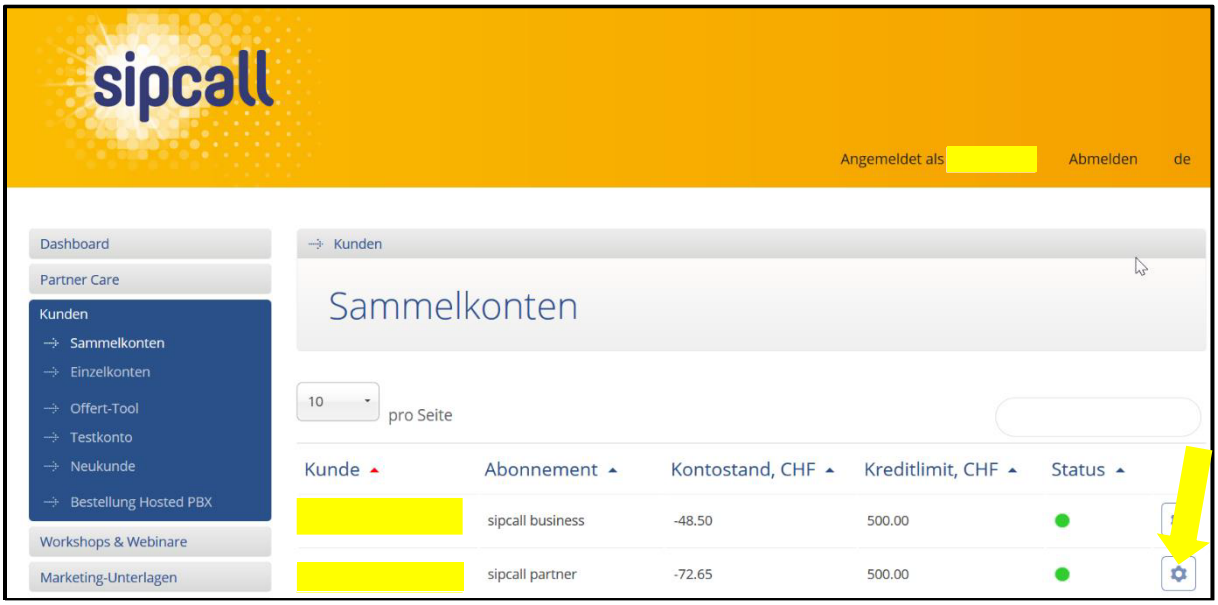



*Figure 8: CDE*

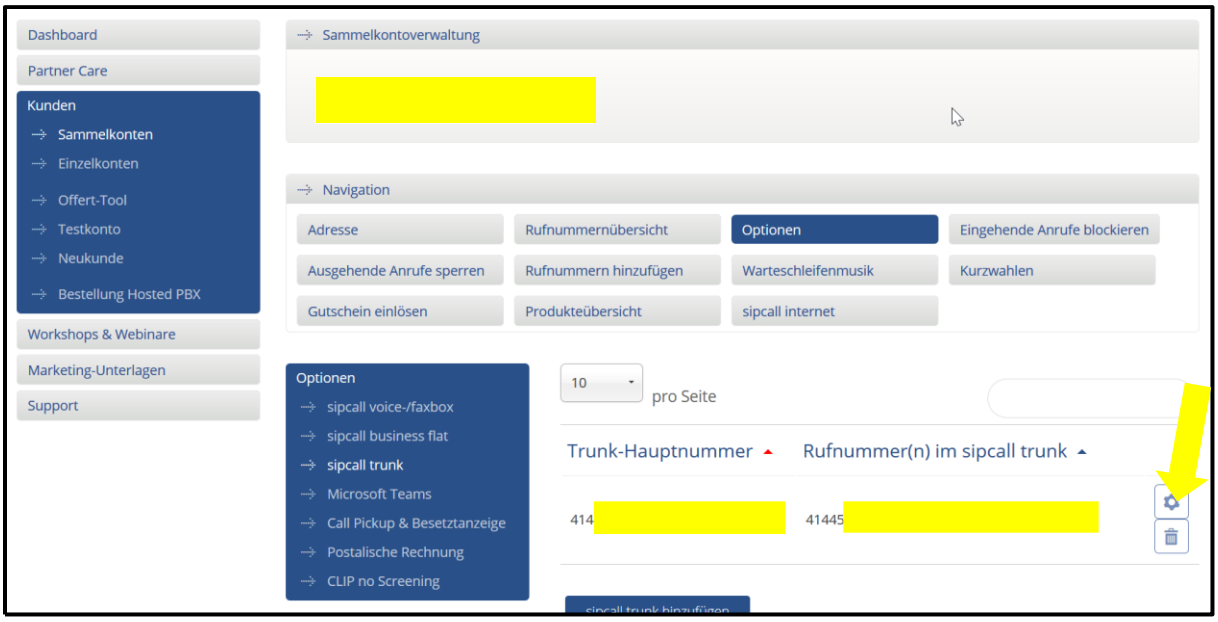
## Provider sipcall

Below are the few screenshots for

### Customer selects




### Option Trunk




**Configuration Trunk**

**Configurations detail for Input in the MiVoice 400**