

Über Mich

- Daniél Kerkmann
- 30 Jahre alt
- Expert Lead, bei OpenTalk seit knapp zwei Jahren
- Zuständig im Bereich Media
- Eine schreiende Katze
- Ein niedliches Kind (aktueller Stand: Kleinkind (1 1/2 Jahre alt))
- Liebe Anime (aktueller Stand: 618 Anime, 181.3 Tage, 11.113 Episoden)
- Rust Enthusiast

Was ist OpenTalk?

The screenshot displays the OpenTalk meeting interface. At the top, the logo "Opentalk" is visible, along with "Meeting-Raum" and a time indicator "13:04". The main video feed shows a woman named Karo. On the left, a sidebar contains a "Breakout Rooms erstellen" (Create Breakout Rooms) menu. A modal window is open over this menu, titled "Laufzeit" (Duration), with options for "Unbegrenzt Zeit" (Unlimited Time), "5 min", "10 min", "15 min", and "30 min", plus a field for "Andere Dauer" (Other Duration). The sidebar also shows "Anzahl der Teilnehmer" (Number of Participants), "zufällige Zuweisung" (Random Assignment), and "inklusive Moderatoren" (Including Moderators) with a toggle switch. A "Räume erstellen" (Create Rooms) button is at the bottom of the sidebar. At the bottom of the screen, a gallery view shows four other participants: Marina, Elias, Sophie, and Patrick.

Wie wir mit Cisco "zusammen" arbeiten

Wie arbeitet OpenTalk mit Cisco zusammen?

Ist es sinnvoll Cisco zu unterstützen?

Welche Probleme traten dabei auf?

Sanfter Einstieg, worum geht es genau?



Quelle: <https://www.best4systems.de/cisco-8945-ip-telefon-general-uuml-berholt>

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Cisco 8945

- Provisionierung über TFTP
- TFTP? (Trivial File Transfer Protocol) Ö.Ö
- XML über TFTP ausgeliefert anhand der MAC.
 - SEP[MAC].cnf.xml

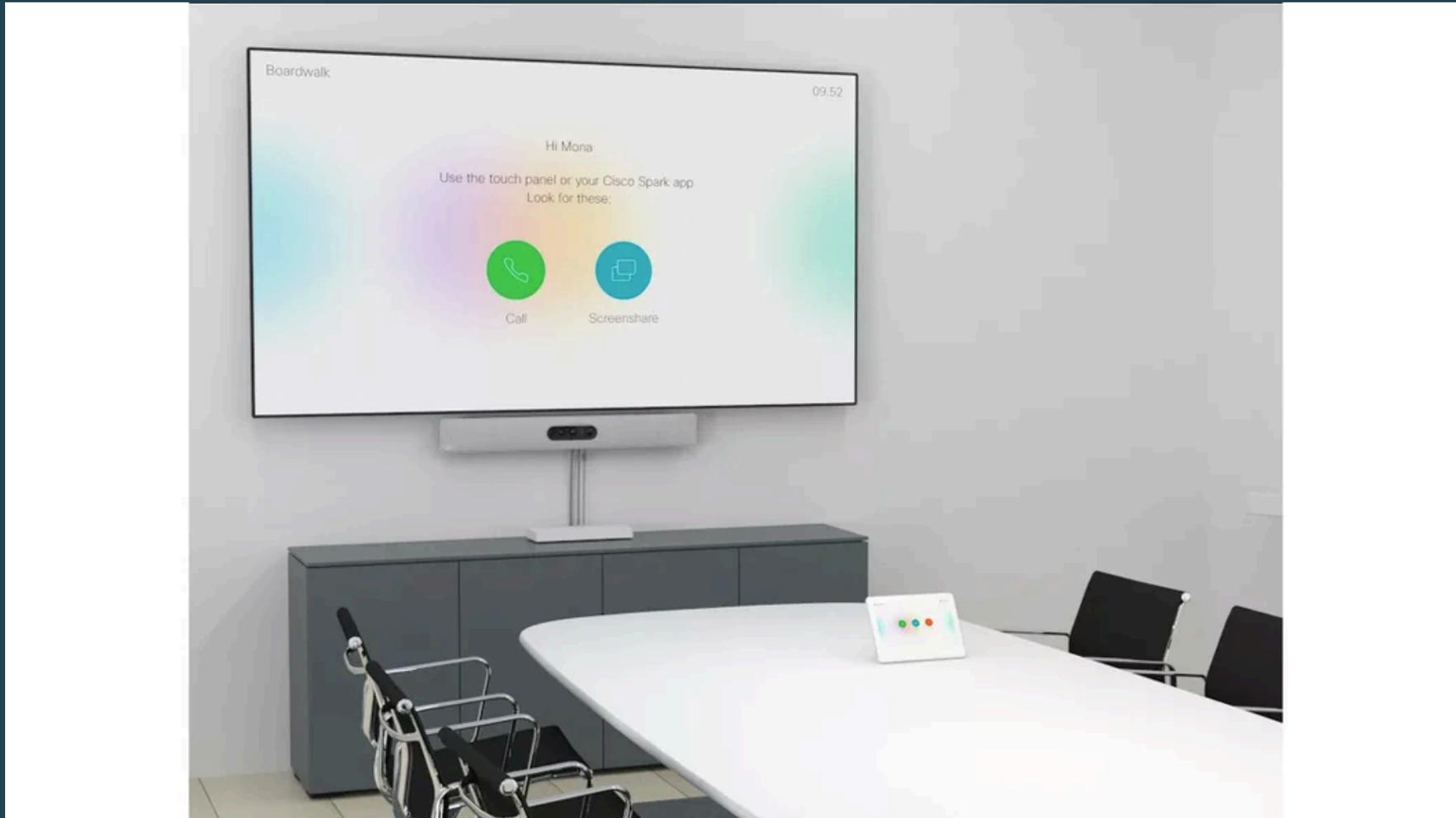
Room Kit



Quelle: <https://www.ipphone-warehouse.com/cisco-room-bar-with-room-navigator-for-table-first-light-cs-bar-t-k9/>

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Room Kit



Quelle: <https://www.dekom.com/de-de/Raumsysteme/Cisco/Cisco-Room-Kit-Pro>

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Room Kit - Dashboard

 10.6.6.113
Cisco Webex DX80

[Home](#) | [Call Control](#) | [Setup](#) | [Security](#) | [Maintenance](#) | [Integration](#) | admin

System Information

There are 5 possible issues with your system. See [Diagnostics](#) for more info.

<h3>General</h3> <p>Product: Cisco Webex DX80 System time: 19:16 Browser time: 19:16 Last boot: 29/01/2025 Serial number: FOC2320N2PU Software version: ce 9.10.0 50f5888d087 2019-12-17 Installed options: Encryption System name: - IPv4: 10.6.6.113 IPv6: - MAC address: 5C:5A:C7:62:CA:B9 Temperature: 41°C / 105.8°F</p>	<h3>H323</h3> <p>Status: Inactive Gatekeeper: - Number: - ID: -</p> <h3>SIP</h3> <p>Status: Failed: Unable to connect to 10.6.6.6:5060 Proxy: 10.6.6.6 URI: 1100@10.6.6.6</p> <p>This video system is not registered</p> <p>In order to place calls with this video system, it needs to be registered to a call service.</p> <p>Register to Webex</p>
---	---

Room Kit - SIP Settings



The screenshot displays the Cisco Webex DX80 configuration interface. The top navigation bar includes 'Home', 'Call Control', 'Setup', 'Security', 'Maintenance', and 'Integration'. The user is logged in as 'admin'. The main content area is titled 'Configuration' and shows the 'SIP' settings page. The left sidebar lists various configuration categories, with 'SIP' currently selected. The SIP settings are organized into several sections:

- SIP Settings:**
 - ANAT: On
 - DefaultTransport: TCP
 - DisplayName: (0 to 550 characters)
 - Line: Shared
 - ListenPort: Off
 - Mailbox: (0 to 255 characters)
 - MinimumTLSVersion: TLSv1.0
 - PreferredPSignaling: IPv4
 - Proxy 1 Address: 10.6.6.6 (0 to 255 characters)
 - Proxy 2 Address: (0 to 255 characters)
 - Proxy 3 Address: (0 to 255 characters)
 - Proxy 4 Address: (0 to 255 characters)
 - TlsVerify: On
 - Type: Cisco
 - URI: 1100@10.6.6.6 (0 to 255 characters)
- Authentication:**
 - Password: (0 to 128 characters)
 - UserName: (0 to 128 characters)
- Ice:**
 - DefaultCandidate: Host
 - Mode: Auto
- Turn:**
 - DiscoverMode: On

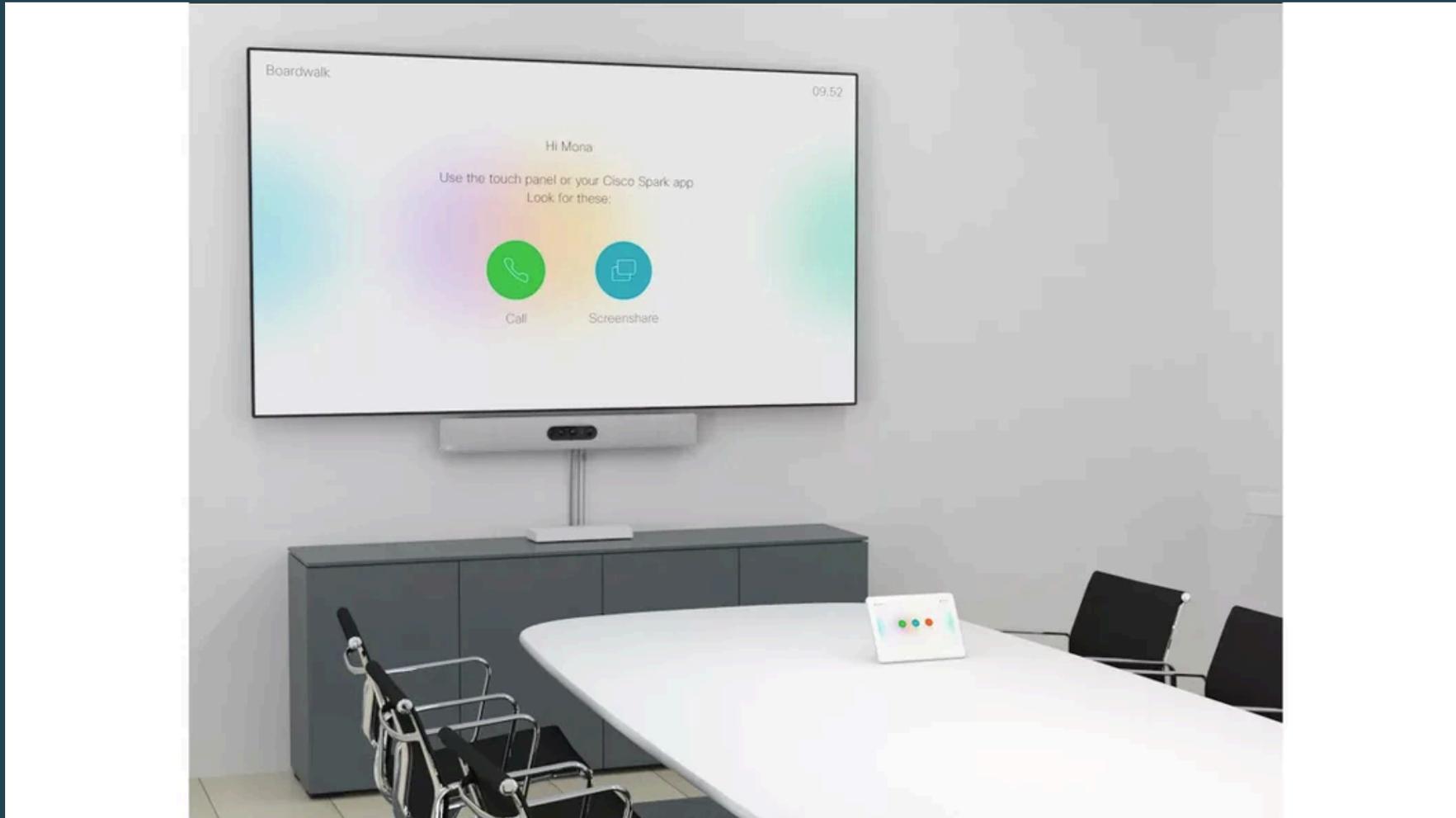


Room Kit - User Interface

The screenshot displays the Cisco Webex DX80 configuration interface. The top navigation bar includes Home, Call Control, Setup, Security, Maintenance, and Integration. The current page is titled "Configuration" and shows the "UserInterface" settings. The left sidebar lists various configuration categories, with "UserInterface" selected. The main content area is divided into sections: "UserInterface", "Features", "Call", "Whiteboard", and "OSD". Each section contains several settings, most of which are dropdown menus or text input fields.

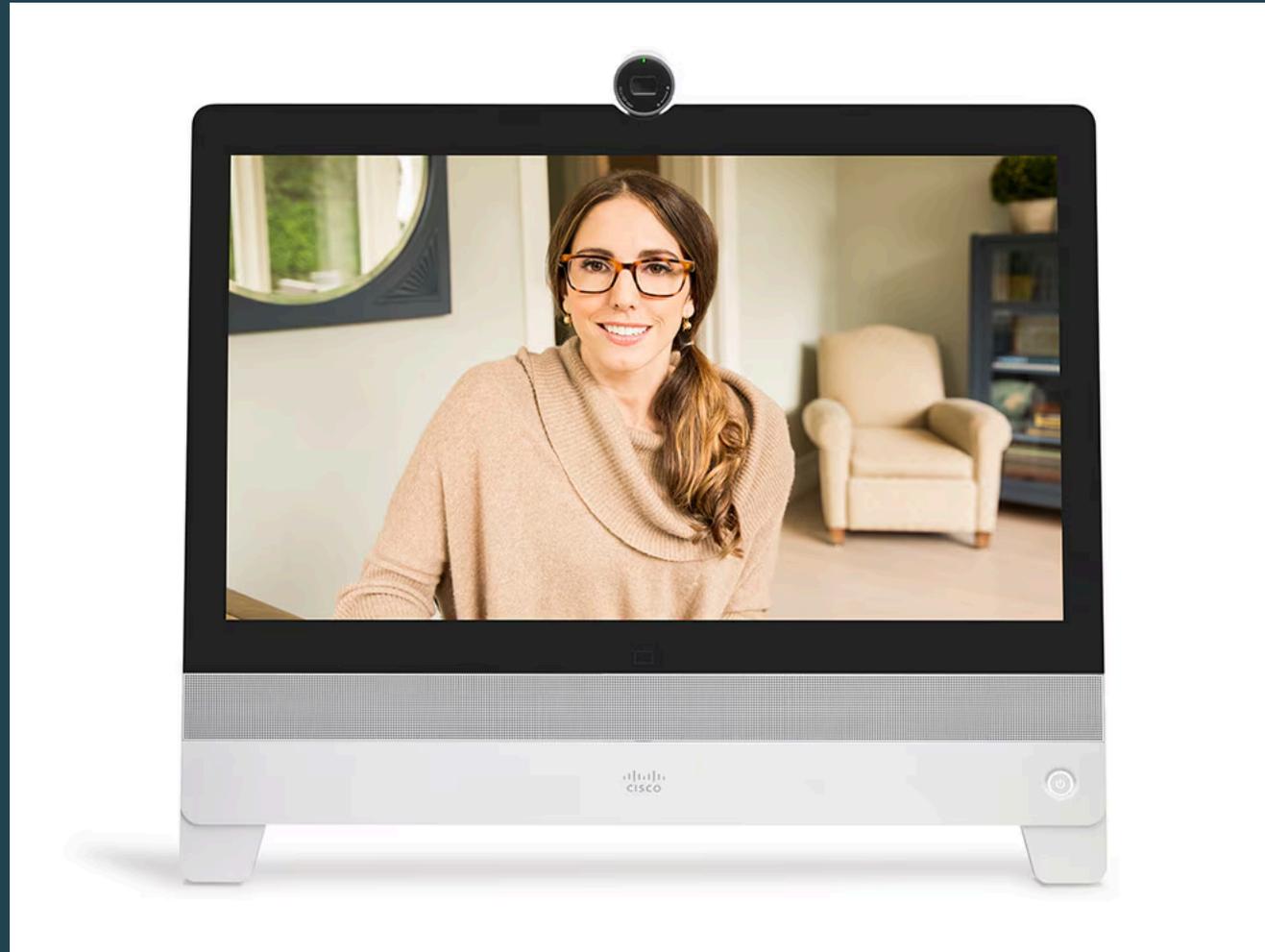
Section	Setting	Value
UserInterface	Accessibility IncomingCallNotification	Default
	Branding AwakeBranding Colors	Auto
	ContactInfo Type	Auto
	CustomMessage	(0 to 128 characters)
	KeyTones Mode	Off
	Language	English
	Phonebook Mode	ReadWrite
	Wallpaper	Auto
Features	HideAll	False
	Share Start	Auto
Call	End	Auto
	JoinWebex	Auto
	MidCallControls	Auto
	Start	Auto
Whiteboard	ExperienceV2	False
	Start	Auto
OSD	EncryptionIndicator	Auto

Room Kit - User Interface



Quelle: <https://www.dekom.com/de-de/Raumsysteme/Cisco/Cisco-Room-Kit-Pro>

Webex DX80



Quelle: https://www.cisco.com/c/de_de/support/collaboration-endpoints/dx80/model.html

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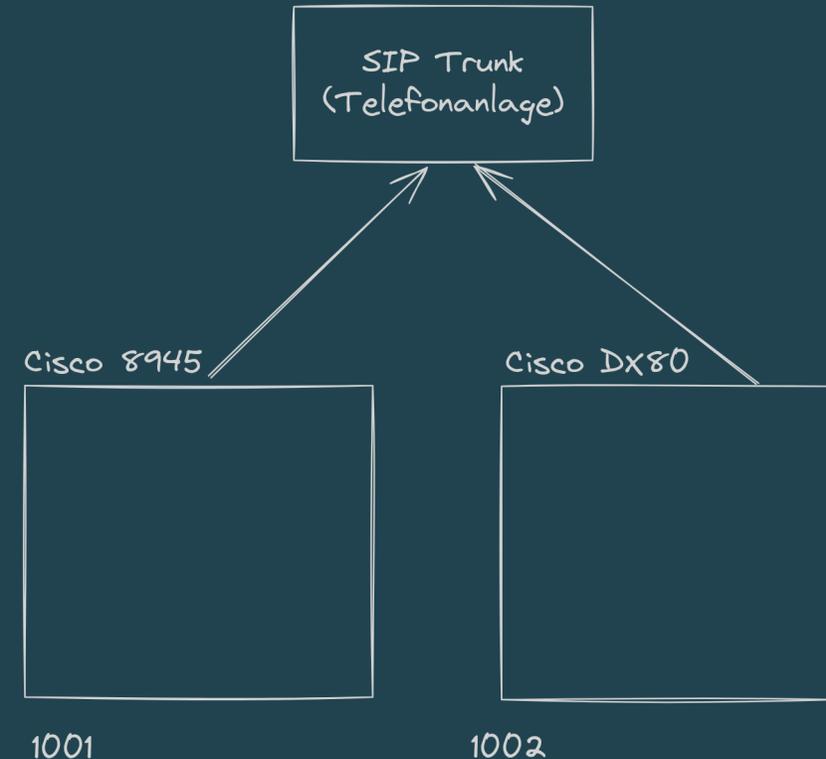
Genug Cisco!

Was ist SIP (Session Initiation Protocol)?



Interne Anrufe

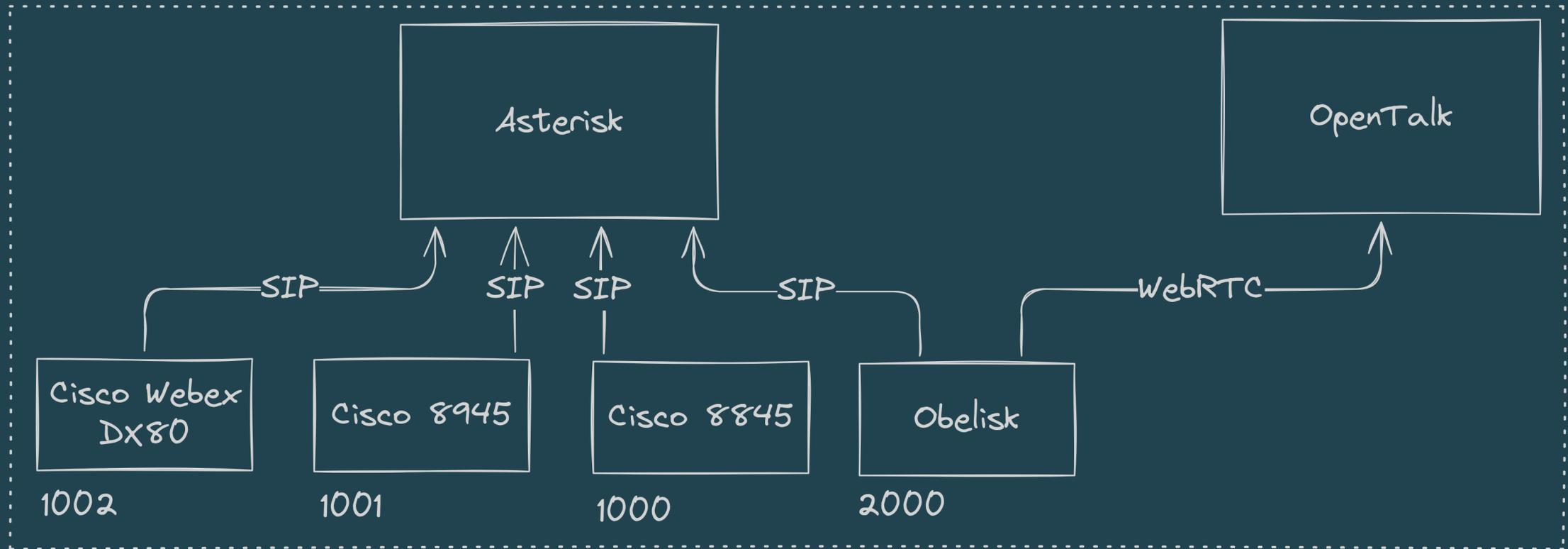
- Cisco 8945: Rufnummer 1001
- Cisco DX80: Rufnummer 1002
- 1001 kann 1002 anrufen (und vice versa)





Wie binden wir OpenTalk an die SIP-Welt an?

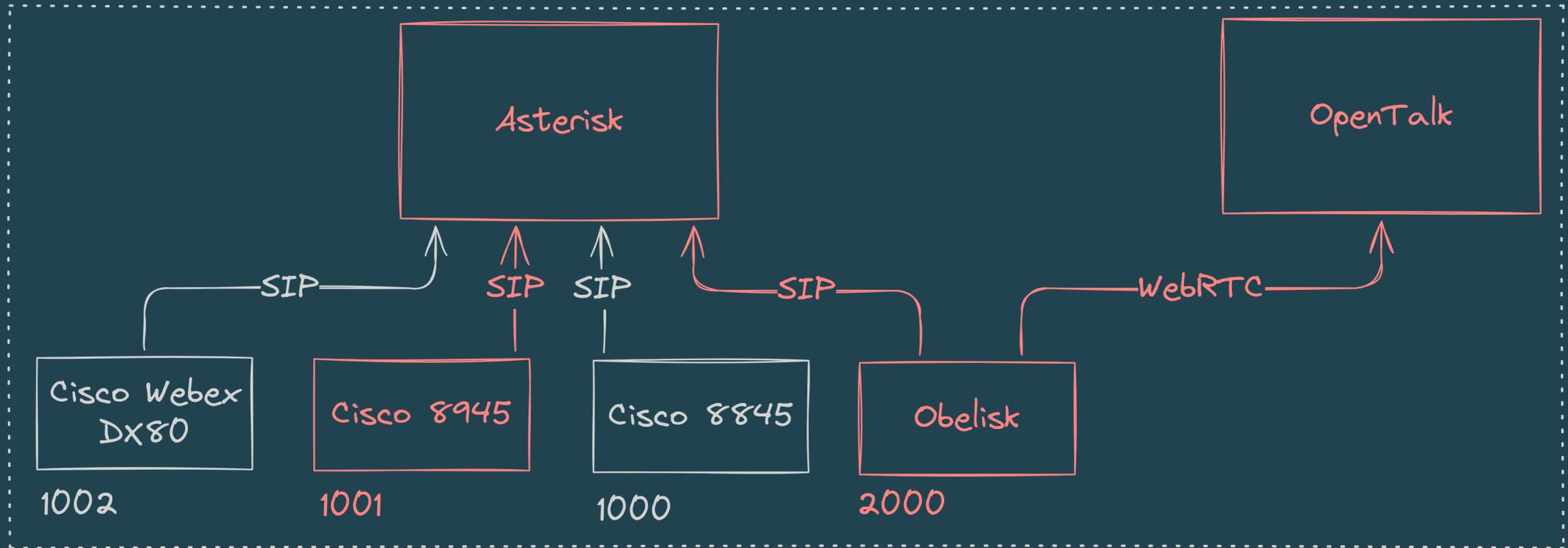
Kommunikation mittels Asterisk





Beispielanruf 1001 -> 2000

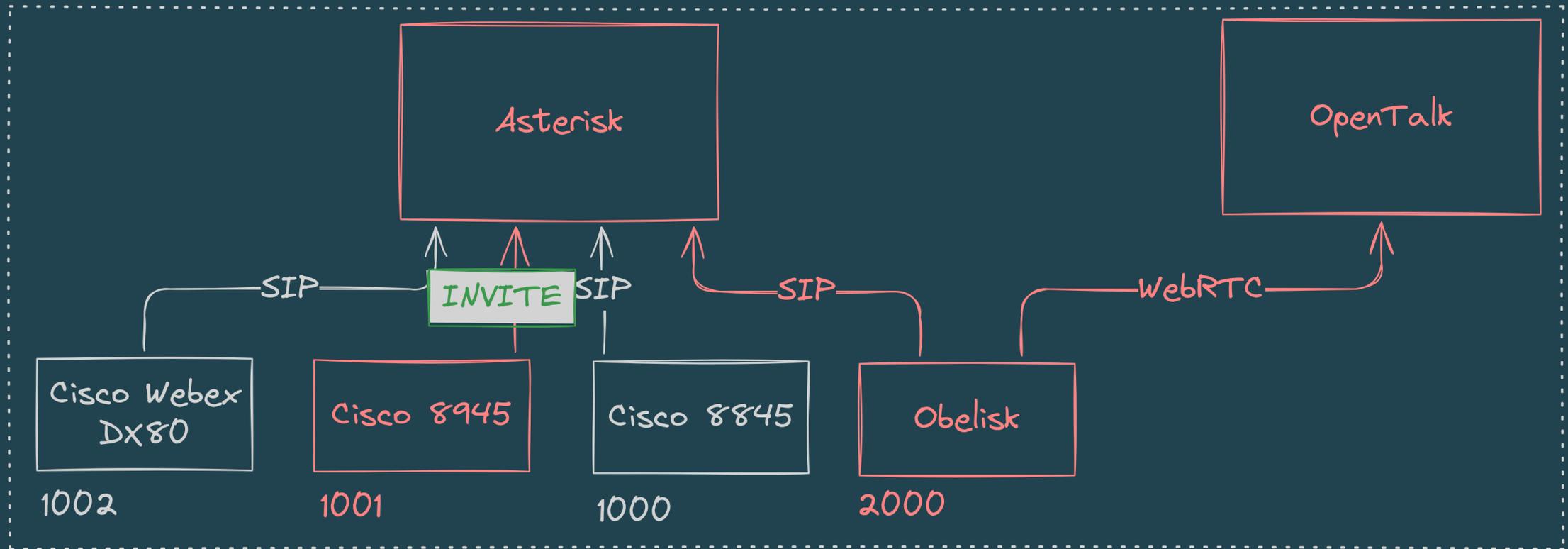
Kommunikation mittels Asterisk





Beispielanruf 1001 -> 2000

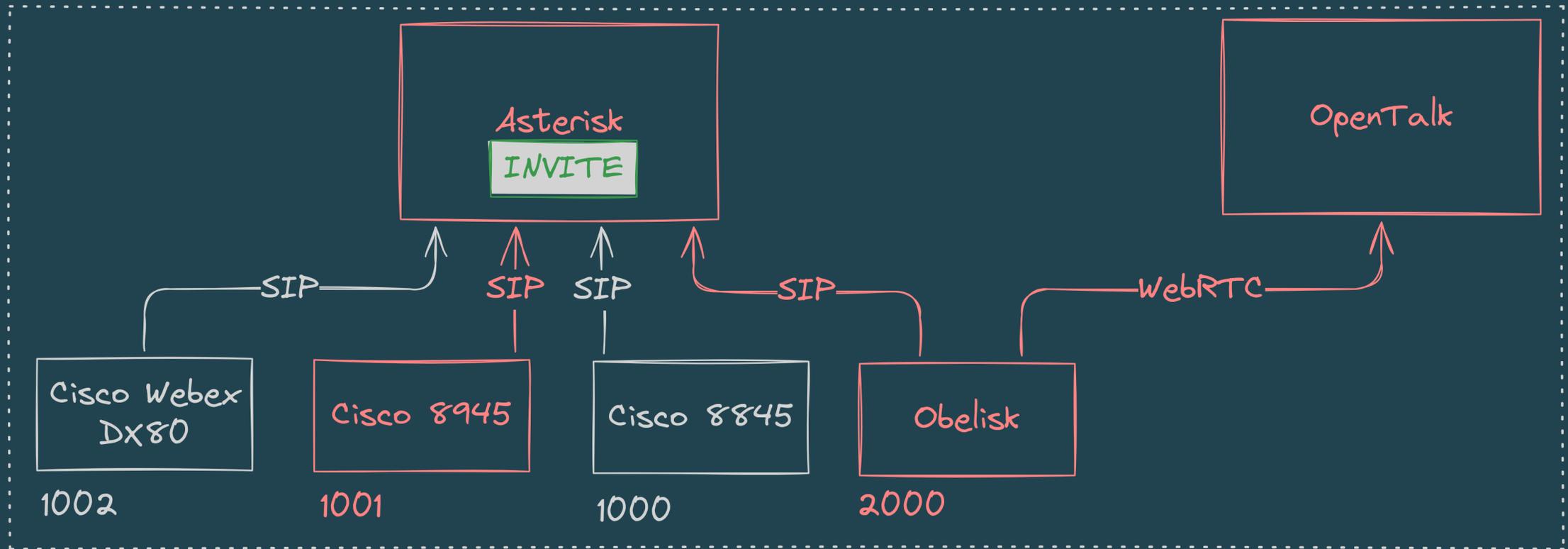
Kommunikation mittels Asterisk





Beispielanruf 1001 -> 2000

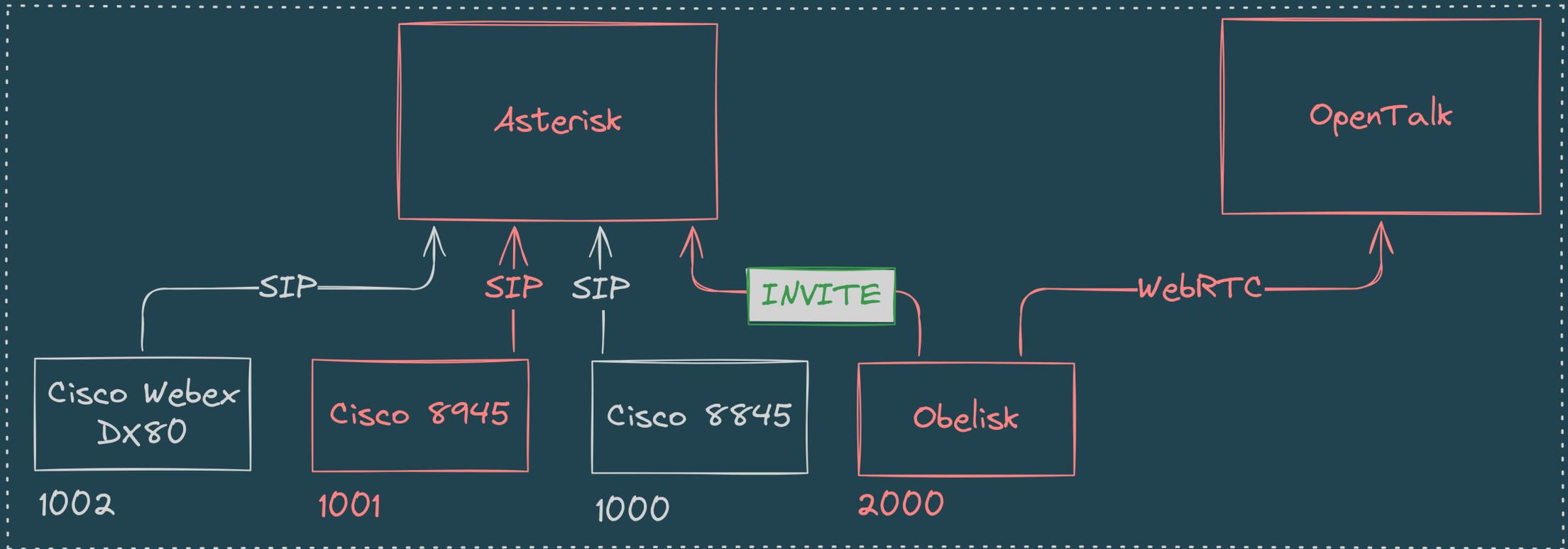
Kommunikation mittels Asterisk





Beispielanruf 1001 -> 2000

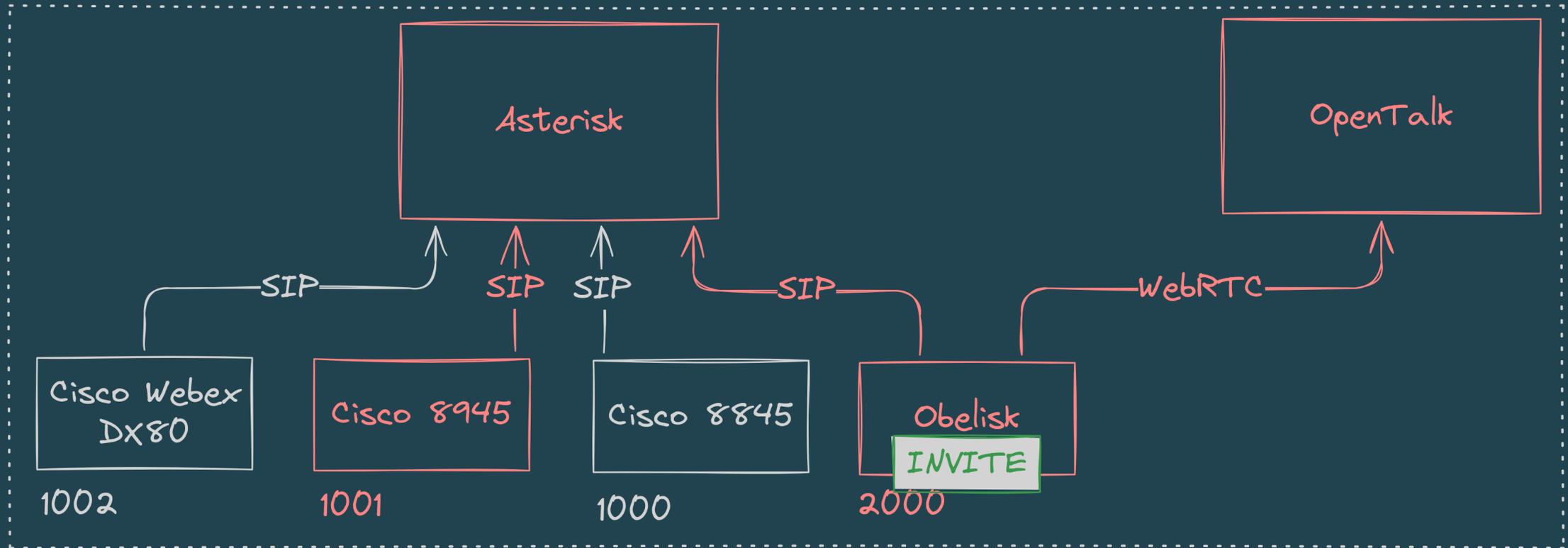
Kommunikation mittels Asterisk





Beispielanruf 1001 -> 2000

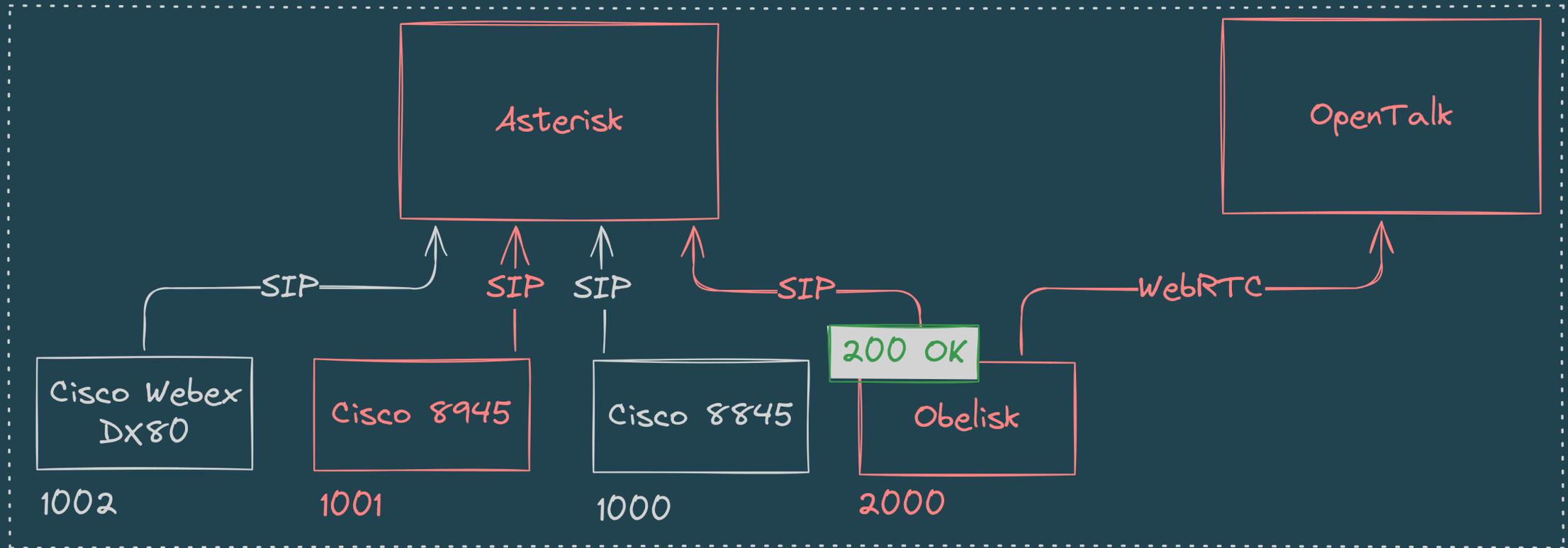
Kommunikation mittels Asterisk





Beispielanruf 1001 -> 2000

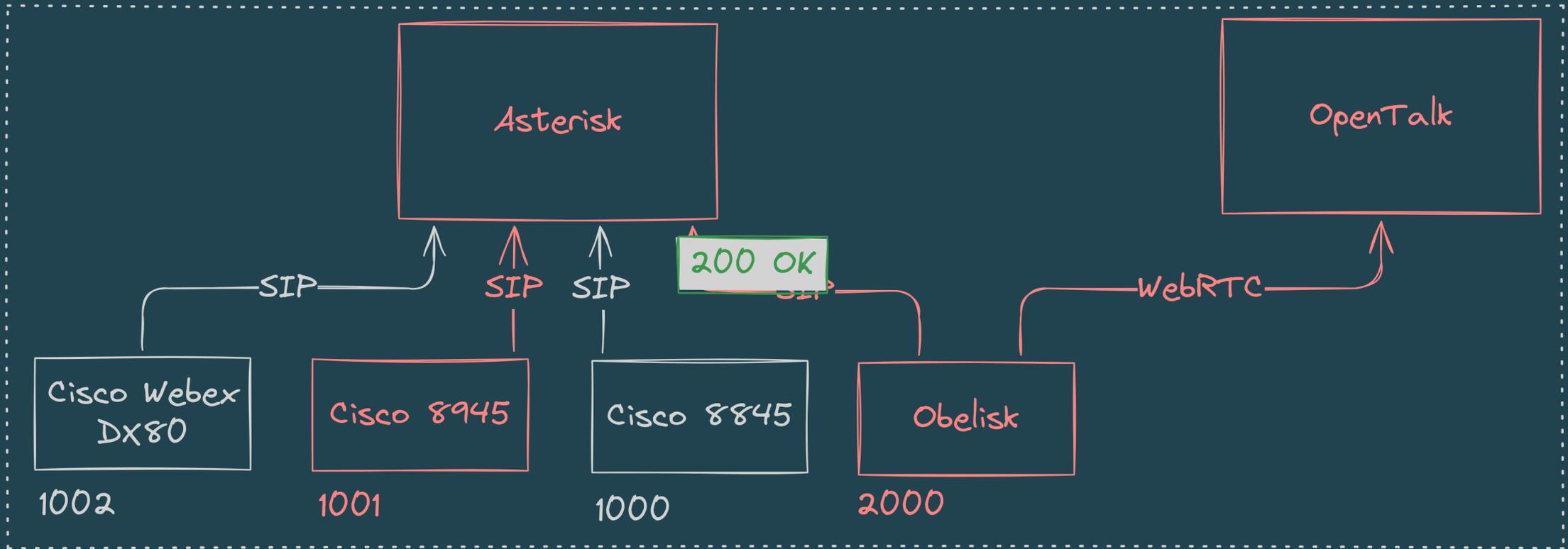
Kommunikation mittels Asterisk





Beispielanruf 1001 -> 2000

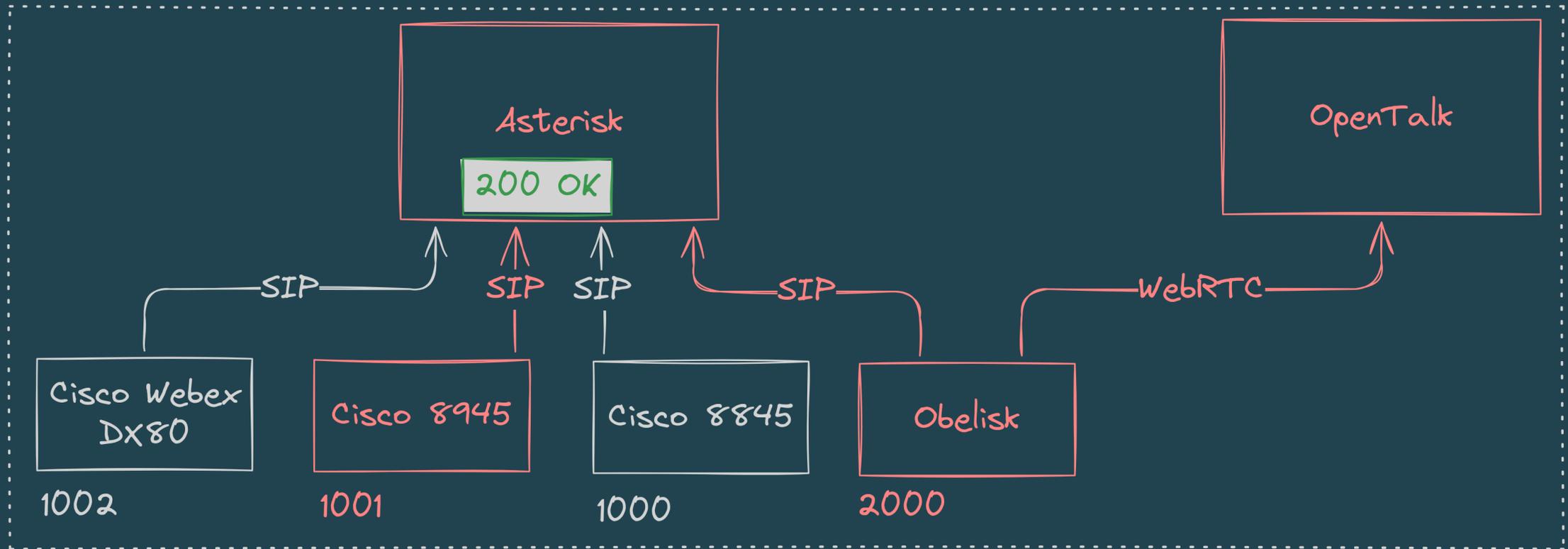
Kommunikation mittels Asterisk





Beispielanruf 1001 -> 2000

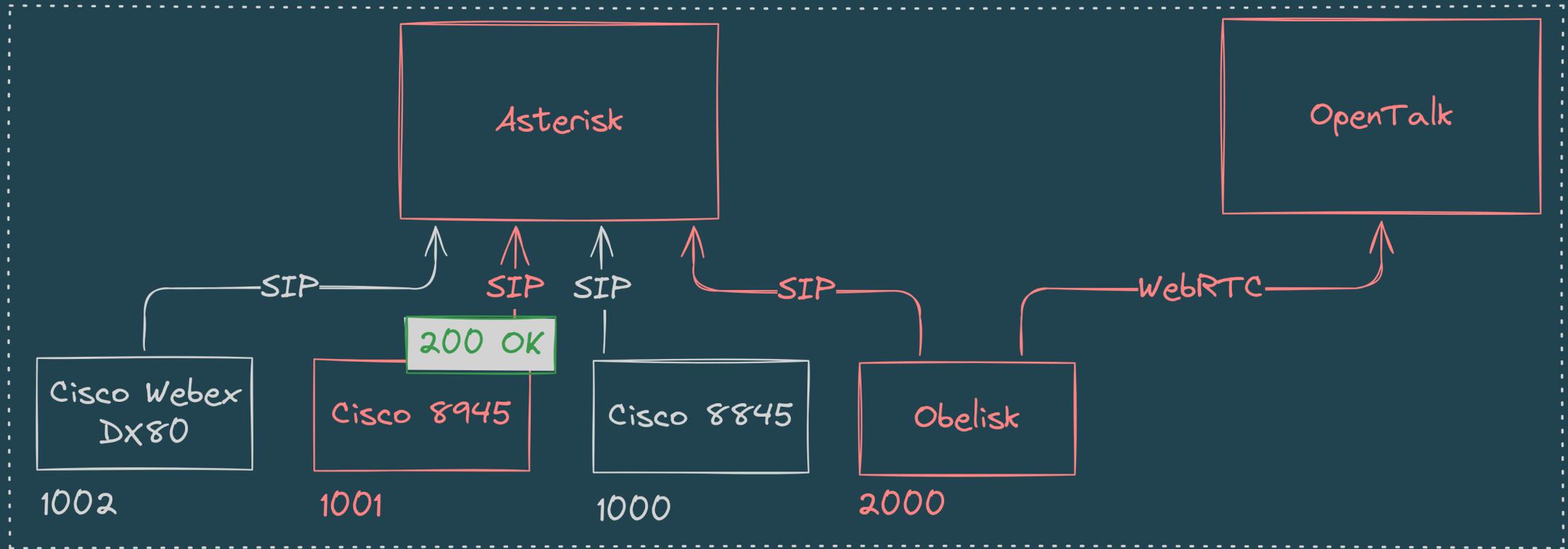
Kommunikation mittels Asterisk





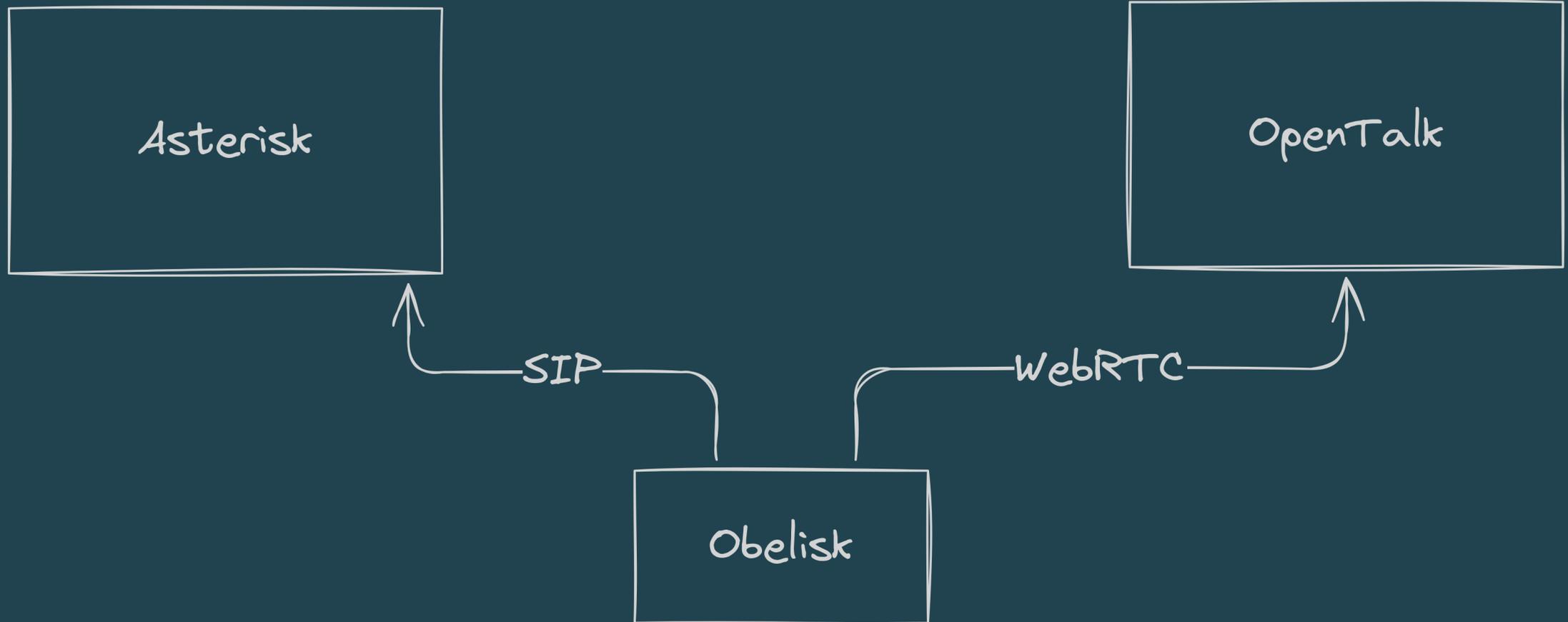
Beispielanruf 1001 -> 2000

Kommunikation mittels Asterisk



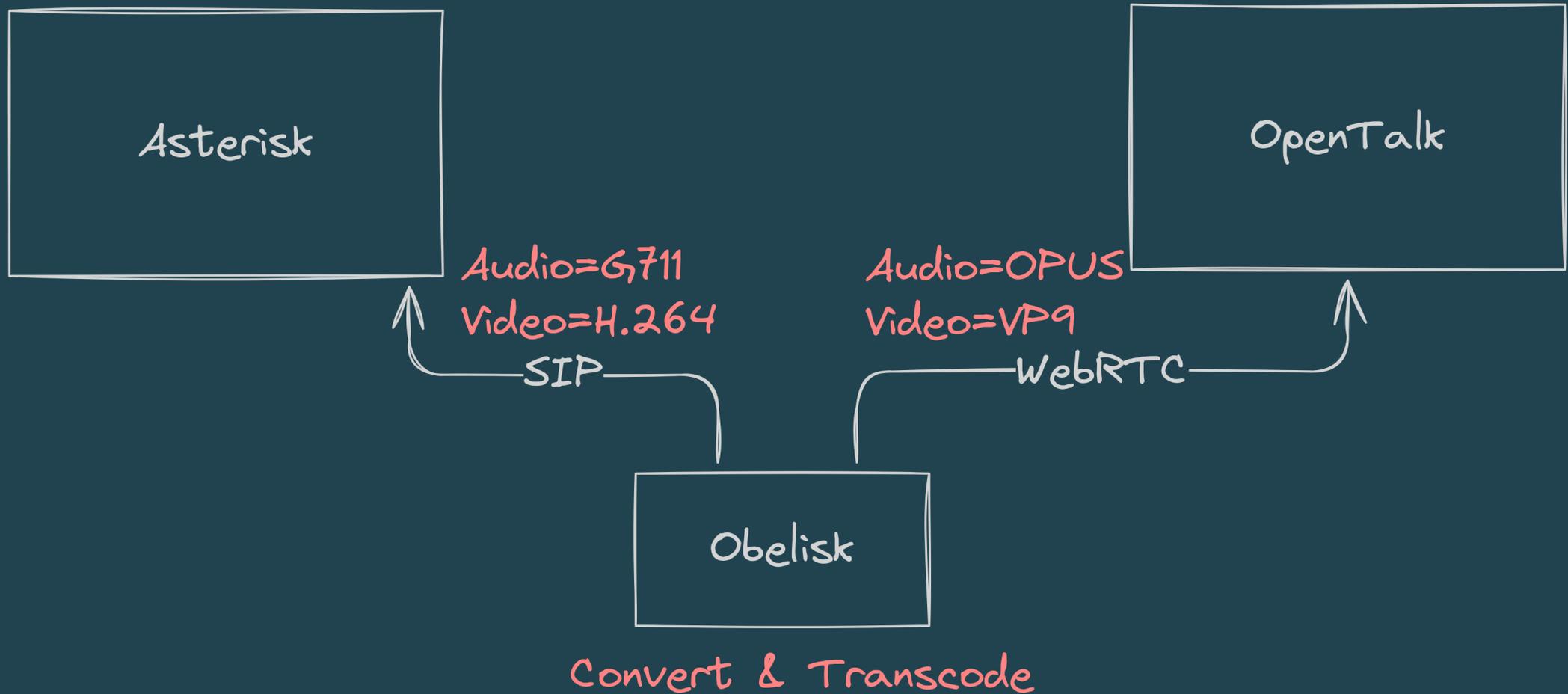


Obelisk im Detail



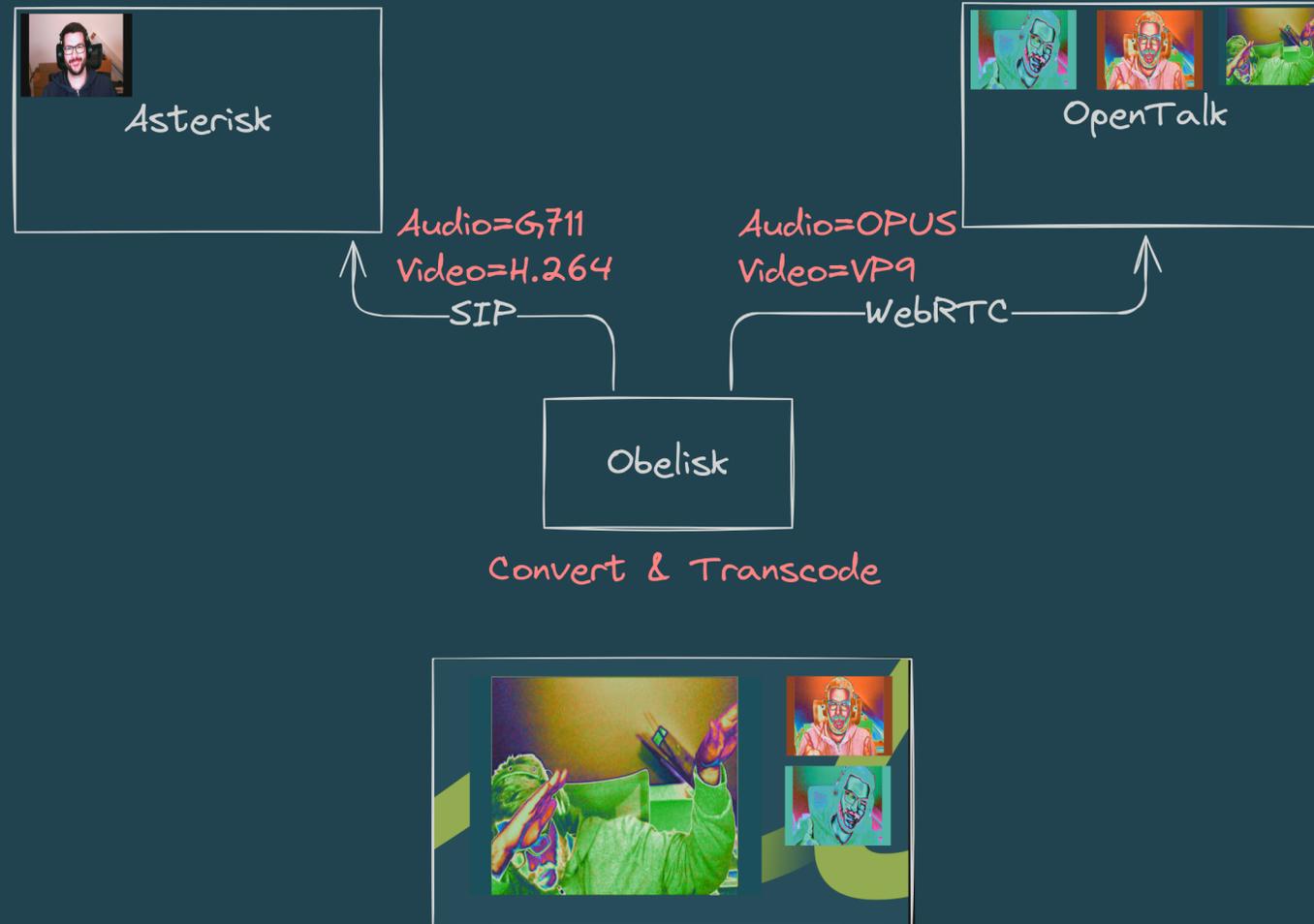


Obelisk im Detail





Obelisk im Detail



Asterix und Obelix?



Quelle: <https://www.britannica.com/topic/Asterix-cartoon-character>



Genug SIP/WebRTC!

Was ist SDP (Session Description Protocol)?

Wie sieht ein SDP-Offer aus?



```
INVITE sip:1001@192.168.3.178:5061 SIP/2.0\r\n
Via: SIP/2.0/UDP 192.168.3.178:5060;rport;branch=z9hG4bKPj46f589cd-a132-4f88-b6da-92f2a70e9541\r\n
From: \"2003@OpenTalk\" <sip:user01@192.168.3.178>;tag=48e31812-96d7-4774-97bc-b468a7d2c721\r\n
To: <sip:1001@192.168.3.178;user=phone>;tag=DQoiUSHQCIJewLWxLJrL50LT4zP\r\n
Contact: <sip:asterisk@192.168.3.178:5060>\r\n
Call-Id: 1fde3c91e801e-439e-a4e9-9d6262e41017\r\n
CSeq: 22777 INVITE\r\n
Allow: OPTIONS, REGISTER, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, INFO, MESSAGE, REFER\r\n
Supported: 100rel, timer, replaces, noferfer, histinfo\r\n
Session-Expires: 1800;refresher=uac\r\n
Min-SE: 90\r\n
Max-Forwards: 70\r\n
User-Agent: OpenTalk VoIP Asterisk Proxy\r\n
Content-Type: application/sdp\r\n
Content-Length: 3264\r\n
\r\n
v=0\r\n
o=- 380173281 380173282 IN IP4 192.168.3.178\r\n
s=Asterisk\r\n
c=IN IP4 192.168.3.178\r\n
t=0 0\r\n
m=audio 14392 RTP/AVP 0 101\r\n
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:a7MmvmvmlIjHeqdTNPkTVFog3jqtQ2WkmsqLGH\r\n
a=ice-ufrag:68ecef744f649fcaeb9a306c8c0875\r\n
a=ice-pwd:5f19aad1128a8a0956a4800b54f93a2f\r\n
a=candidate:Hc0a803b2 1 UDP 2130706431 192.168.3.178 14392 typ host\r\n
a=candidate:Hac110001 1 UDP 2130706431 172.17.0.1 14392 typ host\r\n
a=candidate:Hac170001 1 UDP 2130706431 172.23.0.1 14392 typ host\r\n
a=candidate:Hac150001 1 UDP 2130706431 172.21.0.1 14392 typ host\r\n
a=candidate:Hac120001 1 UDP 2130706431 172.18.0.1 14392 typ host\r\n
a=candidate:Hac130001 1 UDP 2130706431 172.19.0.1 14392 typ host\r\n
a=candidate:Hac140001 1 UDP 2130706431 172.20.0.1 14392 typ host\r\n
a=candidate:Hc0a87a01 1 UDP 2130706431 192.168.122.1 14392 typ host\r\n
a=candidate:Ha003016 1 UDP 2130706431 10.0.48.22 14392 typ host\r\n
a=candidate:Hc0a803b2 2 UDP 2130706430 192.168.3.178 14393 typ host\r\n
a=candidate:Hac110001 2 UDP 2130706430 172.17.0.1 14393 typ host\r\n
a=candidate:Hac170001 2 UDP 2130706430 172.23.0.1 14393 typ host\r\n
a=candidate:Hac150001 2 UDP 2130706430 172.21.0.1 14393 typ host\r\n
a=candidate:Hac120001 2 UDP 2130706430 172.18.0.1 14393 typ host\r\n
a=candidate:Hac130001 2 UDP 2130706430 172.19.0.1 14393 typ host\r\n
a=candidate:Hac140001 2 UDP 2130706430 172.20.0.1 14393 typ host\r\n
a=candidate:Hc0a87a01 2 UDP 2130706430 192.168.122.1 14393 typ host\r\n
a=candidate:Ha003016 2 UDP 2130706430 10.0.48.22 14393 typ host\r\n
a=rtpmap:9 6722/8000\r\n
a=rtpmap:101 telephone-event/8000\r\n
a=fmtp:101 0-16\r\n
a=ptime:20\r\n
a=maxptime:140\r\n
a=sendrecv\r\n
m=video 14184 RTP/AVP 103 99\r\n
a=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:CNi3ShahVvr2rEM59idfVnkL+rRm70kLDiXir100\r\n
a=ice-ufrag:3cc7bc784d150b5f4a7e00b73ed5dcb\r\n
a=ice-pwd:7828925f51a15109720372474984ccc\r\n
a=candidate:Hc0a803b2 1 UDP 2130706431 192.168.3.178 14184 typ host\r\n
a=candidate:Hac110001 1 UDP 2130706431 172.17.0.1 14184 typ host\r\n
a=candidate:Hac170001 1 UDP 2130706431 172.23.0.1 14184 typ host\r\n
a=candidate:Hac150001 1 UDP 2130706431 172.21.0.1 14184 typ host\r\n
a=candidate:Hac120001 1 UDP 2130706431 172.18.0.1 14184 typ host\r\n
a=candidate:Hac130001 1 UDP 2130706431 172.19.0.1 14184 typ host\r\n
a=candidate:Hac140001 1 UDP 2130706431 172.20.0.1 14184 typ host\r\n
a=candidate:Hc0a87a01 1 UDP 2130706431 192.168.122.1 14184 typ host\r\n
a=candidate:Ha003016 1 UDP 2130706431 10.0.48.22 14184 typ host\r\n
a=candidate:Hc0a803b2 2 UDP 2130706430 192.168.3.178 14185 typ host\r\n
a=candidate:Hac110001 2 UDP 2130706430 172.17.0.1 14185 typ host\r\n
a=candidate:Hac170001 2 UDP 2130706430 172.23.0.1 14185 typ host\r\n
a=candidate:Hac150001 2 UDP 2130706430 172.21.0.1 14185 typ host\r\n
a=candidate:Hac120001 2 UDP 2130706430 172.18.0.1 14185 typ host\r\n
a=candidate:Hac130001 2 UDP 2130706430 172.19.0.1 14185 typ host\r\n
a=candidate:Hac140001 2 UDP 2130706430 172.20.0.1 14185 typ host\r\n
a=candidate:Hc0a87a01 2 UDP 2130706430 192.168.122.1 14185 typ host\r\n
a=candidate:Ha003016 2 UDP 2130706430 10.0.48.22 14185 typ host\r\n
a=rtpmap:103 h263-1998/90000\r\n
a=fmtp:103 CIF4=1,CIF=1,QCIF=1,CUSTOM=352,240,1;MaxBr=30000\r\n
a=rtpmap:99 H264/90000\r\n
a=fmtp:99 max-mbps=245000;max-fs=8160;max-br=2500;max-smbps=245000;max-fps=3000;packetization-mode=0;profile-level-id=428014\r\n
a=sendrecv\r\n
```



Wie sieht ein SDP-Offer aus? (Meta)

```
INVITE sip:1001@192.168.3.178:5061 SIP/2.0\r\n
Via: SIP/2.0/UDP 192.168.3.178:5060;rport;branch=z9hG4bKPj46f589cd-a132-4f88-b6da-92f2a70e9541\r\n
From: \"2003@OpenTalk\" <sip:user01@192.168.3.178>;tag=48e31812-96d7-4774-97bc-b460a7d2c721\r\n
To: <sip:1001@192.168.3.178;user=phone>;tag=DQoiUShQCIJewLtWhXLljLr50LT4zp\r\n
Contact: <sip:asterisk@192.168.3.178:5060>\r\n

Call-ID: 1fde3c91-981e-439e-a4e9-9d6262e41017\r\n
CSeq: 22777 INVITE\r\n

Allow: OPTIONS, REGISTER, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, INFO, MESSAGE, REFER\r\n
Supported: 100rel, timer, replaces, norefersub, histinfo\r\n
Session-Expires: 1800;refresher=uac\r\n
Min-SE: 90\r\n
Max-Forwards: 70\r\n
User-Agent: OpenTalk VoIP Asterisk Proxy\r\n
Content-Type: application/sdp\r\n
Content-Length: 3264\r\n

v=0\r\n
o=- 380173281 380173282 IN IP4 192.168.3.178\r\n
s=Asterisk\r\n
c=IN IP4 192.168.3.178\r\n
t=0 0\r\n
```



Wie sieht ein SDP-Offer aus? (Audio)

```
m=audio 14392 RTP/AVP 9 101\r\n\na=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:a7MwmqvmulIjHeqdTNPKTVFoq3jqTq2WkmsgqLGH\r\n\na=ice-frag:68ecef744f649fc0aeb9a306c8c0875\r\na=ice-pwd:5f19aad1128a8a0956a4800b54f93a2f\r\na=candidate:Hc0a803b2 1 UDP 2130706431 192.168.3.178 14392 typ host\r\na=candidate:Hac110001 1 UDP 2130706431 172.17.0.1 14392 typ host\r\n...\na=candidate:Hc0a87a01 2 UDP 2130706430 192.168.122.1 14393 typ host\r\na=candidate:Ha003016 2 UDP 2130706430 10.0.48.22 14393 typ host\r\n\na=rtpmap:9 G722/8000\r\n\na=rtpmap:101 telephone-event/8000\r\na=fmtp:101 0-16\r\naptime:20\r\na=maxptime:140\r\n\na=sendrecv\r\n
```



Wie sieht ein SDP-Offer aus? (Video)

```
m=video 14184 RTP/AVP 103 99\r\n\r\na=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:CNI3haHVYr2rIME59idfVnkL+rRm70kldGIx1r00\r\n\r\na=ice-ufrag:3cc7bc704d150b5f4a71ed0b73ed5dcb\r\na=ice-pwd:7626025f51a15160720372474984ccccf\r\na=candidate:Hc0a803b2 1 UDP 2130706431 192.168.3.178 14184 typ host\r\na=candidate:Hac110001 1 UDP 2130706431 172.17.0.1 14184 typ host\r\n\r\n...\r\na=candidate:Hc0a87a01 2 UDP 2130706430 192.168.122.1 14185 typ host\r\na=candidate:Ha003016 2 UDP 2130706430 10.0.48.22 14185 typ host\r\n\r\na=rtpmap:103 h263-1998/90000\r\na=fmtp:103 CIF4=1;CIF=1;QCIF=1;CUSTOM=352,240,1;MaxBR=30000\r\n\r\na=rtpmap:99 H264/90000\r\na=fmtp:99 max-mbps=245000;max-fs=8160;max-br=2500;max-smbps=245000;max-fps=3000;packetization-mode=0;profile-level-id=428014\r\n\r\na=sendrecv\r\n
```

What about H.264?



315 / 812 | - 165% + | [] ↻

Table A-1 – Level limits

Level number	Max macroblock processing rate MaxMBPS (MB/s)	Max frame size MaxFS (MBs)	Max decoded picture buffer size MaxDpbMbs (MBs)	Max video bit rate MaxBR (1000 bits/s, 1200 bits/s, cpbBrVclFactor bits/s, or cpbBrNalFactor bits/s)	Max CPB size MaxCPB (1000 bits, 1200 bits, cpbBrVclFactor bits, or cpbBrNalFactor bits)	Vertical MV component limit MaxVmvR (luma frame samples)	Min compression ratio MinCR	Max number of motion vectors per two consecutive MBs MaxMvsPer2Mb
1	1 485	99	396	64	175	64	2	-
1b	1 485	99	396	128	350	64	2	-
1.1	3 000	396	900	192	500	128	2	-
1.2	6 000	396	2 376	384	1 000	128	2	-
1.3	11 880	396	2 376	768	2 000	128	2	-
2	11 880	396	2 376	2 000	2 000	128	2	-
2.1	19 800	792	4 752	4 000	4 000	256	2	-
2.2	20 250	1 620	8 100	4 000	4 000	256	2	-
3	40 500	1 620	8 100	10 000	10 000	256	2	32
3.1	108 000	3 600	18 000	14 000	14 000	512	4	16
3.2	216 000	5 120	20 480	20 000	20 000	512	4	16
4	245 760	8 192	32 768	20 000	25 000	512	4	16
4.1	245 760	8 192	32 768	50 000	62 500	512	2	16
4.2	522 240	8 704	34 816	50 000	62 500	512	2	16
5	589 824	22 080	110 400	135 000	135 000	512	2	16
5.1	983 040	36 864	184 320	240 000	240 000	512	2	16
5.2	2 073 600	36 864	184 320	240 000	240 000	512	2	16
6	4 177 920	139 264	696 320	240 000	240 000	8 192	2	16
6.1	8 355 840	139 264	696 320	480 000	480 000	8 192	2	16
6.2	16 711 680	139 264	696 320	800 000	800 000	8 192	2	16

Quelle: Recommendation ITU-T H.264

Wie sieht die SDP-Answer aus?



```
SIP/2.0 200 OK\r\n
Via: SIP/2.0/UDP 192.168.3.178:5060;rport=5060;branch=z9hG4bKPj46f589cd-a132-4f88-b6da-92f2a70e9541\r\n
From: \"2003@OpenTalk\"<sip:user01@192.168.3.178>;tag=48e31812-96d7-4774-97bc-b460a7d2c721\r\n
To: <sip:1001@192.168.3.178;user=phone>;tag=DQoiUShQCIJewLtWhXLljLr50LT4zp\r\n

Call-ID: 1fde3c91-981e-439e-a4e9-9d6262e41017\r\n
CSeq: 22777 INVITE\r\n

Contact: <sip:1001@192.168.3.178:5061>\r\n

Allow: INVITE, UPDATE, BYE, ACK, CANCEL, PRACK\r\n
Supported: 100rel, timer\r\n
Content-Type: application/sdp\r\n
Content-Length: 333\r\n

v=0\r\n
o=- 1733237008 1733237009 IN IP4 192.168.3.178\r\n
s=opentalk-obelisk\r\n
t=0 0\r\n

m=audio 43734 RTP/AVP 9 101\r\n
c=IN IP4 192.168.3.178\r\n
a=sendrecv\r\n
a=rtcp:43735\r\n
a=rtpmap:9 G722/8000\r\n
a=rtpmap:101 telephone-event/8000\r\n
a=fmtp:101 0-16\r\n

m=video 46808 RTP/AVP 99\r\n
c=IN IP4 192.168.3.178\r\n
a=sendrecv\r\n
a=rtcp:46809\r\n
a=rtpmap:99 H264/90000\r\n
```



Genug Erklärungen!

Demo mit Asterisk.



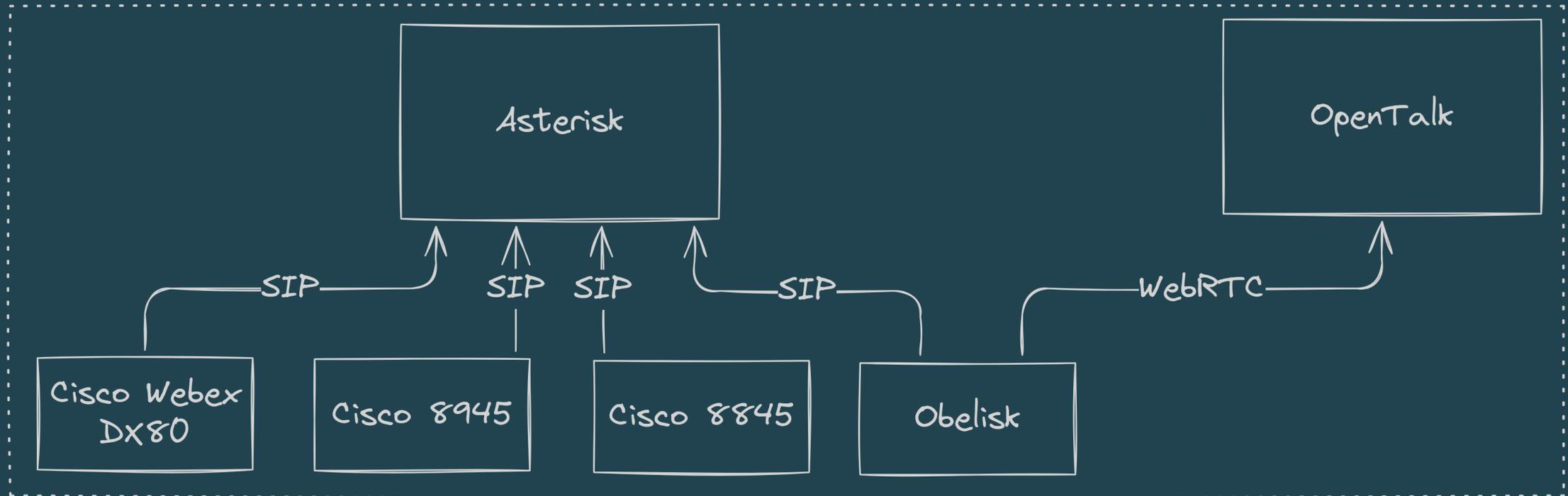
Encoding läuft gut :^)





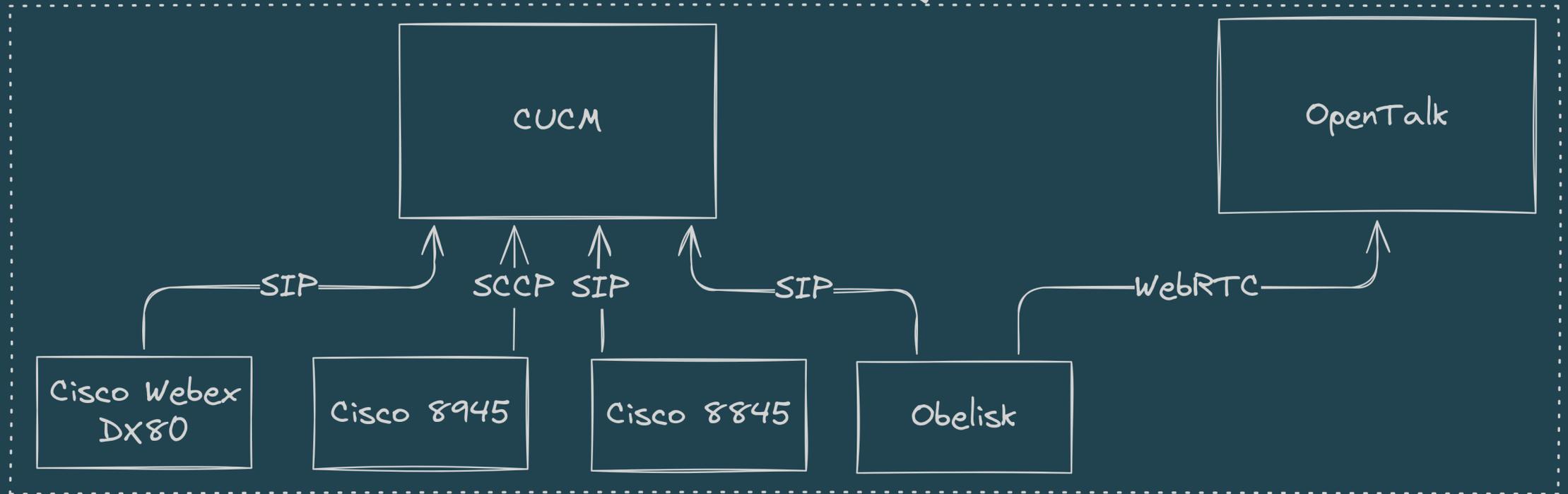
Rückblick wie Asterisk funktioniert

Kommunikation mittels Asterisk

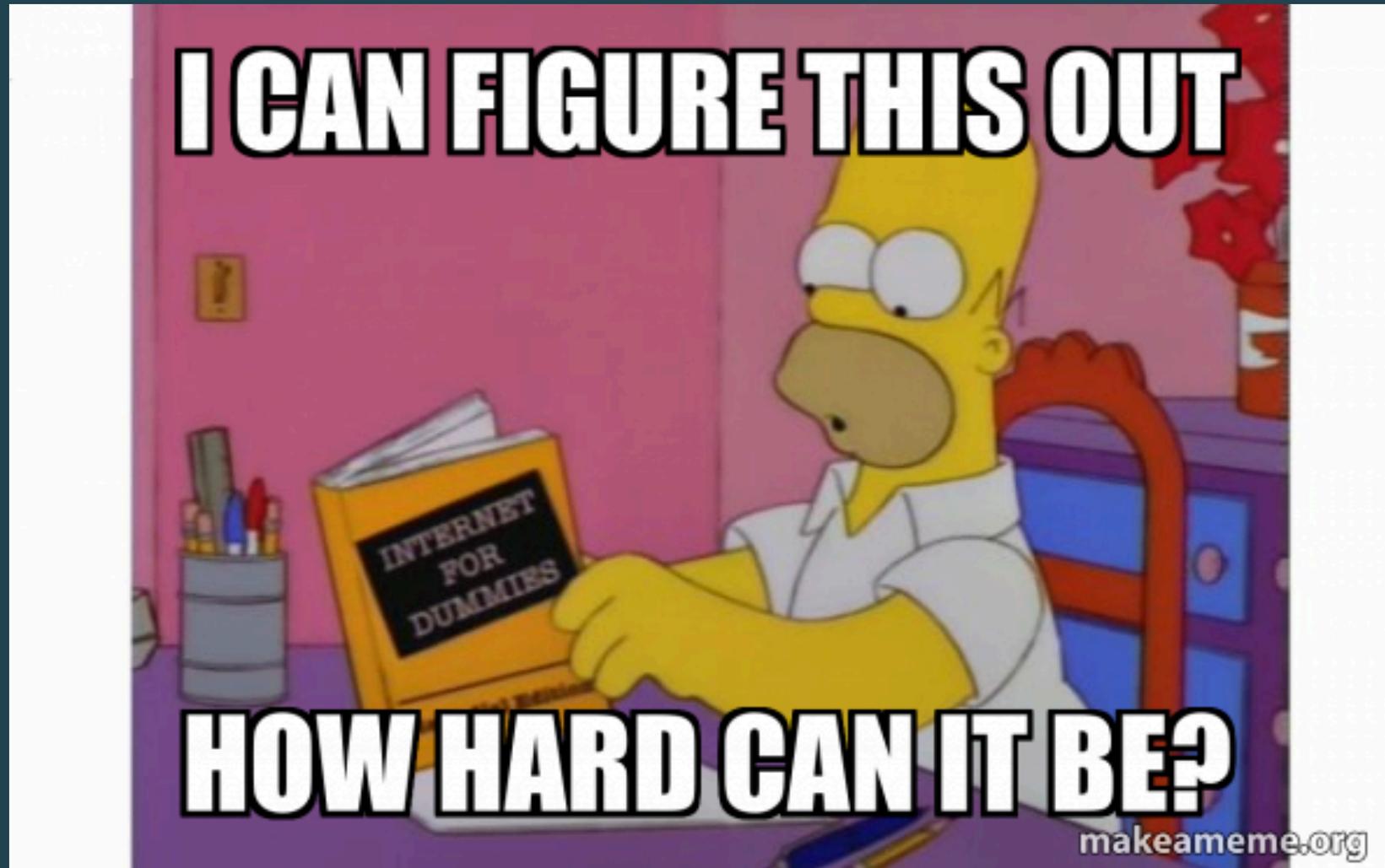


Einfach den Asterisk durch Cisco ersetzen, right?

Kommunikation mittels Cisco Unified Communications Manager (CUCM)



Cisco installieren, wie schwierig kann es sein ...



Cisco installieren, wie schwierig kann es sein ...



- Schritt 1: Cisco Unified Communications Manager downloaden

Wie Sie Cisco Lösungen kaufen Partner Anmelden DE DE

Produkte und Services Lösungen Support Lernen Informationen zu Cisco Suchen

Support / Produkt-Support / Unified Communications /

Cisco Unified Communications Manager (CallManager)

Überblick Produktübersicht
Produkttyp Anrufsteuerung
Status Verfügbar **Bestellung aus Serie**
Veröffentlichungsdatum der Serie Pre-1999

Cisco kontaktieren Englisch

Versionen Dokumentation Downloads Community Zurückgezogene Versionen

- Unified Communications Manager Version 15**
Status: Verfügbar | Veröffentlichungsdatum: 16-Oct-2023
- Unified Communications Manager Version 14**
Status: Verfügbar | Veröffentlichungsdatum: 24-Mar-2021
- Unified Communications Manager Version 12.5**
Status: Nicht bestellbar | End-of-Support-Datum: 31-Aug-2025
- Unified Communications Manager Version 12.0**
Status: Support-Ende | End-of-Support-Datum: 31-Aug-2023
- Unified Communications Manager Version 11.5**
Status: Support-Ende | End-of-Support-Datum: 31-May-2024

Unternehmensinfo Kontakt Karriere bei Cisco Mit einem Partner verbinden

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- Möchten Sie Cisco Partner werden?



Die Lösung

The screenshot shows the Internet Archive website interface. At the top, there is a navigation bar with the Internet Archive logo and various media type filters: WEB, TEXTS, VIDEO, AUDIO, SOFTWARE, and IMAGES. To the right of these filters are links for SIGN UP | LOG IN and an UPLOAD button, followed by a search bar. Below the navigation bar is a secondary menu with links for ABOUT, BLOG, PROJECTS, HELP, DONATE (with a red heart icon), CONTACT, JOBS, VOLUNTEER, and PEOPLE.

The main content area displays the title "Files for cisco-unified-communications-manager-12.5.1.13900". Below this title is a table listing the files in the directory. The table has three columns: Name, Last modified, and Size.

Name	Last modified	Size
Go to parent directory		
CUCM/	19-Mar-2021 18:37	-
cisco-unified-communications-manager-12.5.1.13900_archive.torrent	19-Mar-2021 22:48	45.8K
cisco-unified-communications-manager-12.5.1.13900_files.xml	09-Oct-2024 01:40	2.2K
cisco-unified-communications-manager-12.5.1.13900_meta.sqlite	19-Mar-2021 18:37	21.0K
cisco-unified-communications-manager-12.5.1.13900_meta.xml	19-Mar-2021 22:48	1.4K



Virtuelle Maschine angeschmissen und los gehts!

```
Detecting Server Hardware - this can take several minutes  
Detecting Server Hardware - this can take several minutes  
Detecting Server Hardware - this can take several minutes  
Detecting Server Hardware - this can take several minutes
```



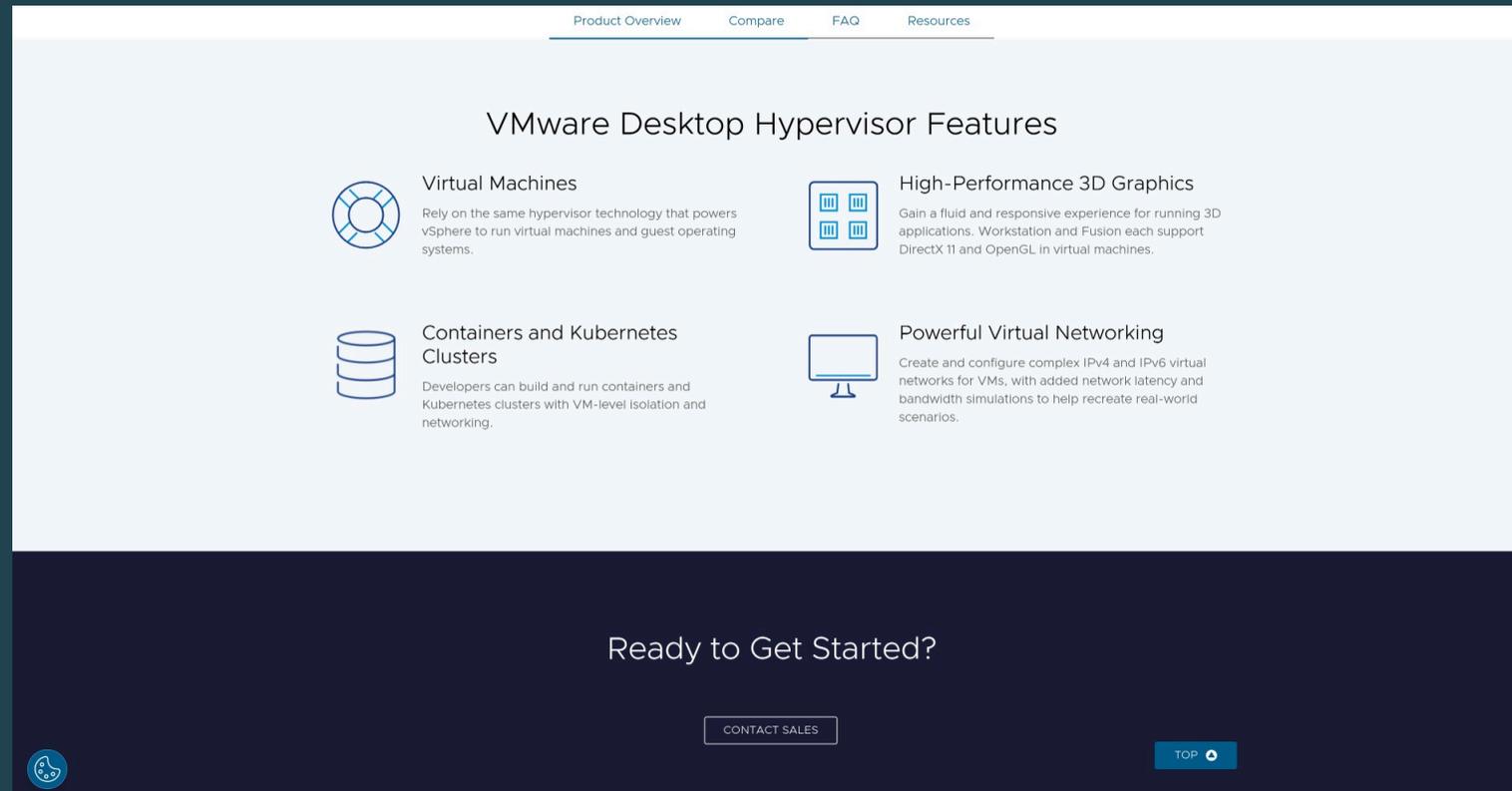
Critical Error

The hardware you are using is not supported for this product.
Installation will now halt.

Continue

VMWare installieren, wie schwierig kann es sein ..

- Schritt 1: VMWare downloaden



- Möchten Sie VMWare Partner werden?



Die Lösung

INTERNET ARCHIVE

WEB TEXTS VIDEO AUDIO SOFTWARE IMAGES

SIGN UP | LOG IN UPLOAD Search

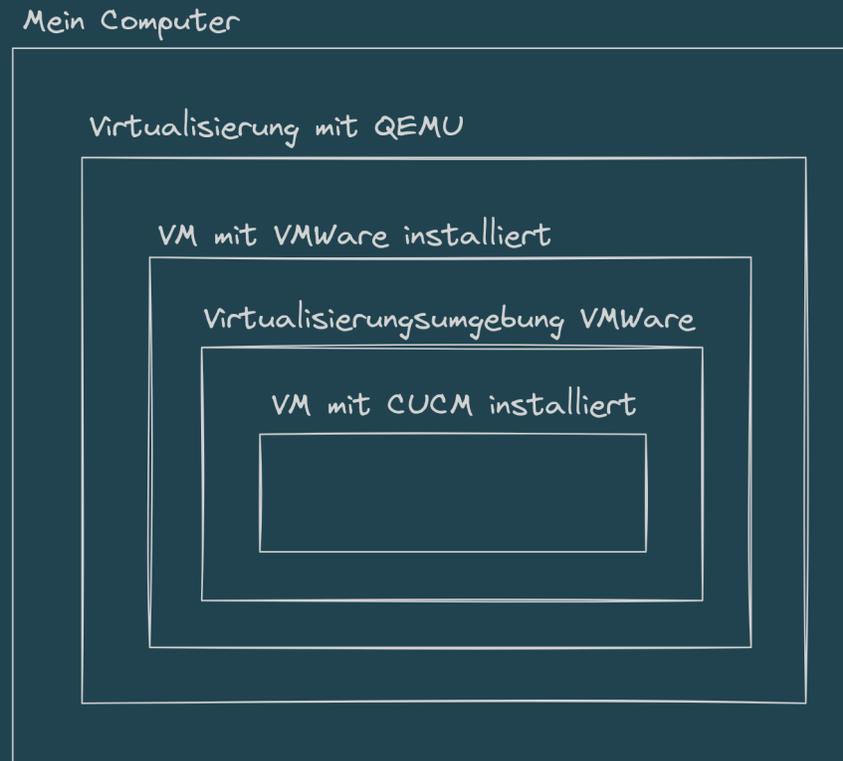
ABOUT BLOG PROJECTS HELP DONATE CONTACT JOBS VOLUNTEER PEOPLE

Files for vmware-esxi-7

Name	Last modified	Size
Go to parent directory		
Vmware-VMvisor-Installer-7.0U3f-20036589.x86_64.iso (View Contents)	01-Sep-2022 09:07	383.0M
__ia_thumb.jpg	01-Sep-2022 09:08	4.7K
esxi.jpg	01-Sep-2022 09:08	12.7K
esxi_thumb.jpg	01-Sep-2022 09:09	6.2K
vmware-esxi-7_archive.torrent	26-Aug-2024 21:19	17.1K
vmware-esxi-7_files.xml	26-Aug-2024 21:19	2.4K
vmware-esxi-7_meta.sqlite	01-Sep-2022 09:08	20.0K
vmware-esxi-7_meta.xml	25-Aug-2024 17:55	907.0B



Unser aktuelles Setup





VM-Ception

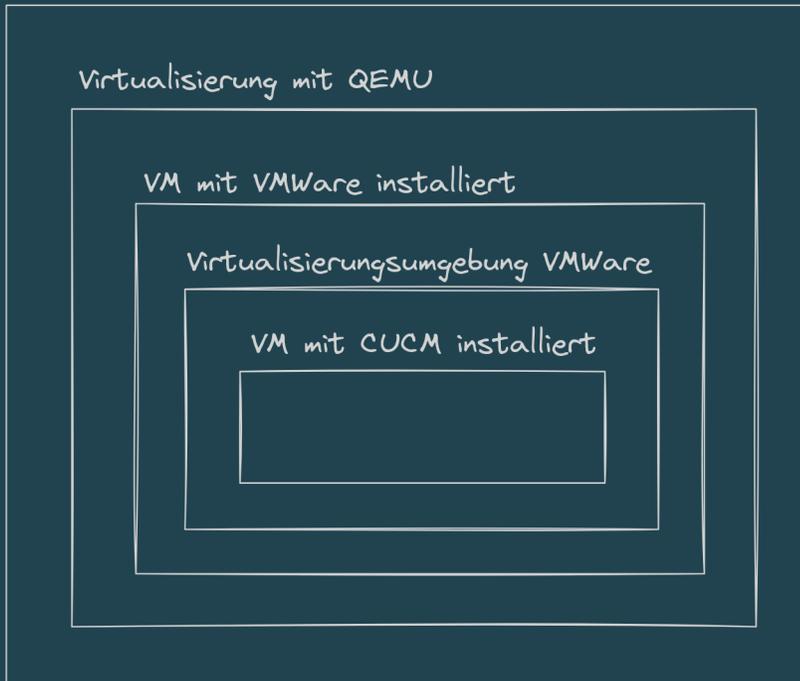


Quelle: <https://www.linkedin.com/pulse/running-vm-container-ramesh-kumar>

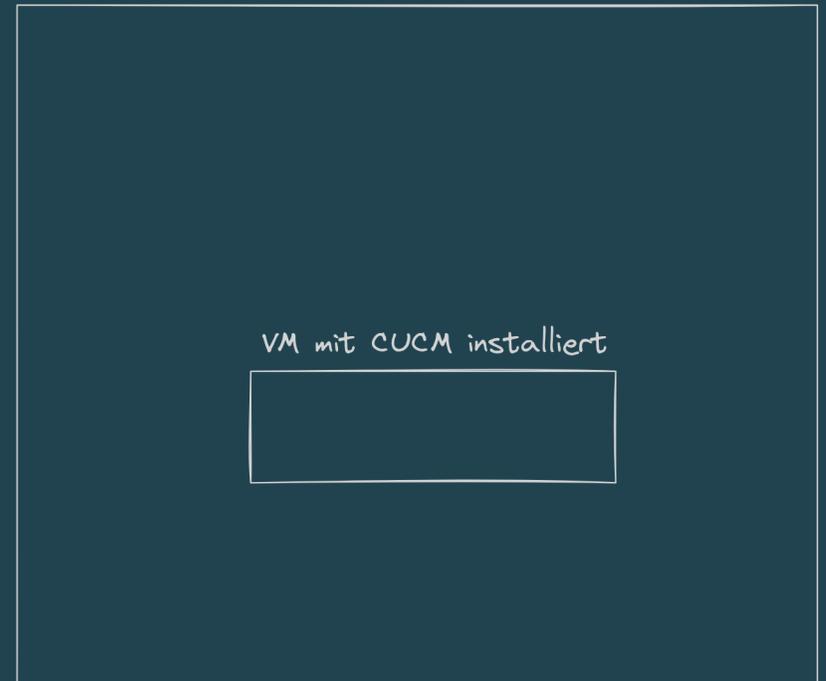


Unser neuestes neuestes Setup (Note)

Mein Computer



Synology NAS





Now we got it ... Let's install



NTP Error?!

Cisco Unified Communications Manager 15.0.1.11901-2

Error

The NTP server(s) are mistyped, inaccessible or unreliable.

Verify that valid NTP server names or IP addresses were entered, that they are running NTPv4 at stratum 6 or less, and that port 123 is not blocked by a firewall.

OK



Checking Network

- 5 Minuten Netzwerk-check
- Network check timeout exceeded
- Network check timeout exceeded, extending timeout by 20 minutes



Finally, ein Terminal!

```
Command Line Interface is starting up, please wait ...
```



Erstmal eine statische IP

```
admin:set network ip eth0 ?
```

```
Syntax:
```

```
set network ip eth0 addr mask
```

```
addr      mandatory    IP addr to be assigned
```

```
mask      mandatory    IP mask to be assigned
```

```
gw        mandatory    IP default gw to be assigned
```

```
admin:set network ip eth0 _
```



Erstmal eine statische IP einstellen

Erstmal 5 Minuten nichts



Erstmal eine statische IP einstellen

Database corrupted

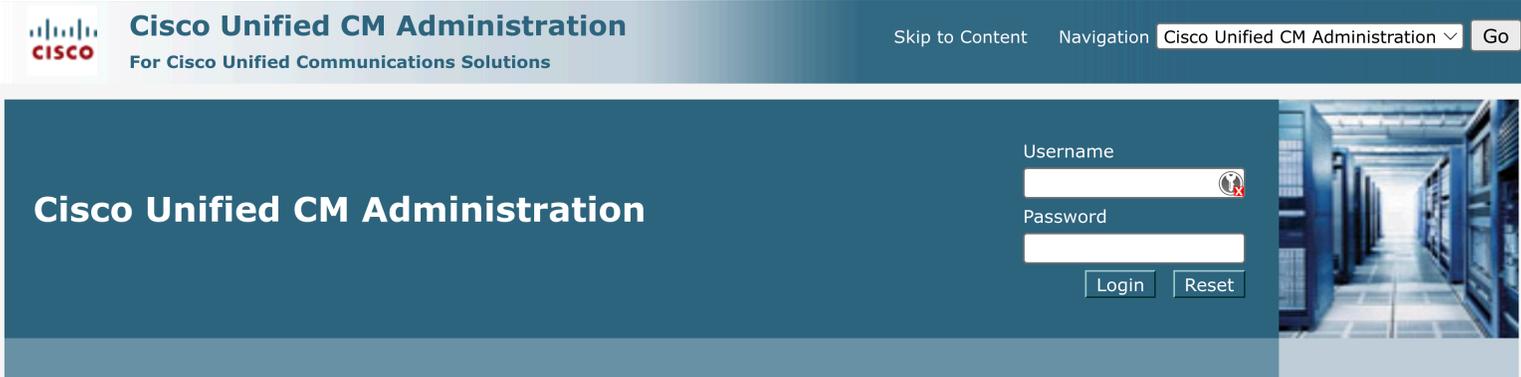


Die Lösung

- CUCM neu installieren
- DHCP-Server sagen welche IP die MAC bekommen soll
- Installation dauert ja nur mehrere Stunden :^)



Finally, eine WebUI!



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Skip to Content Navigation **Cisco Unified CM Administration**

Cisco Unified CM Administration

Username

Password

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This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

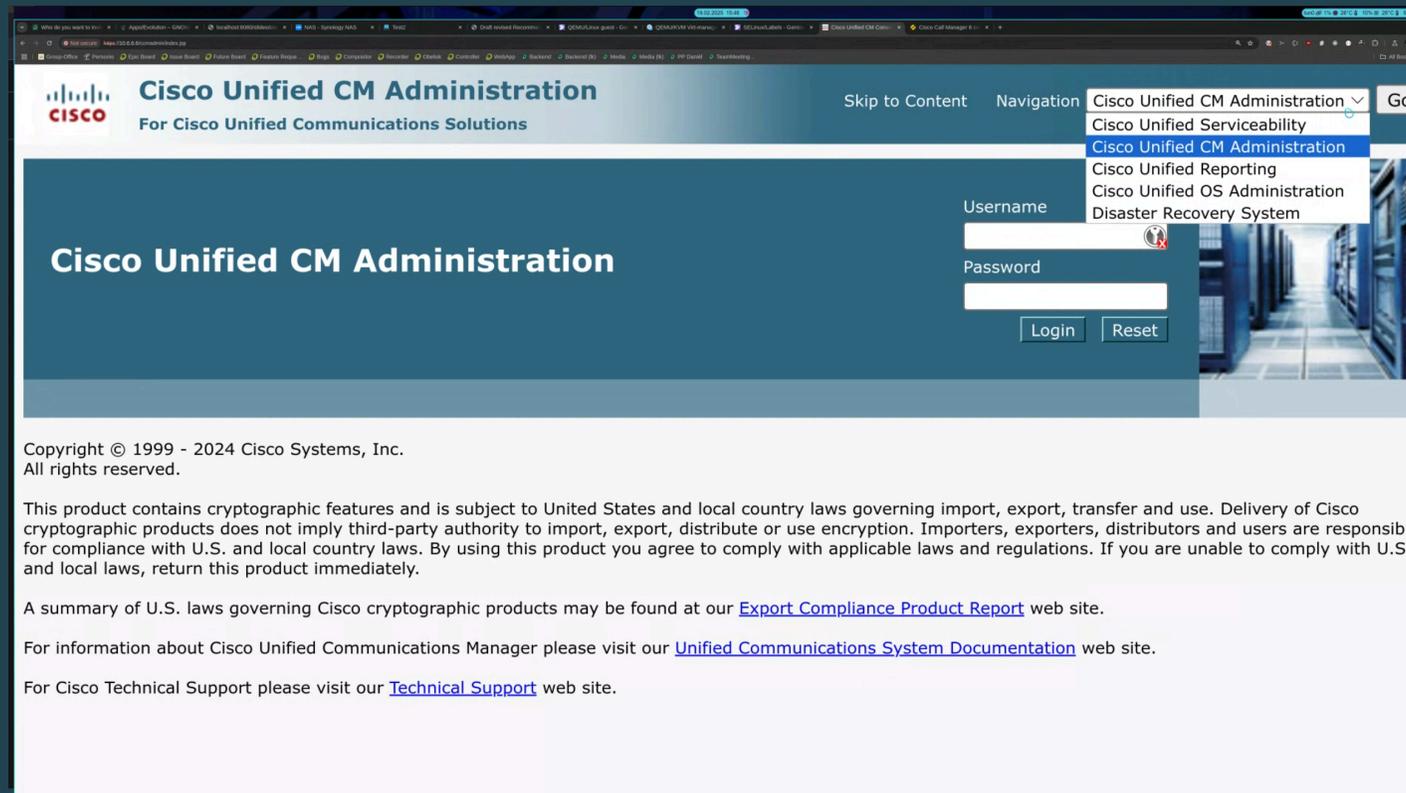
A summary of U.S. laws governing Cisco cryptographic products may be found at our [Export Compliance Product Report](#) web site.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.



Finally, "EINE" WebUI?



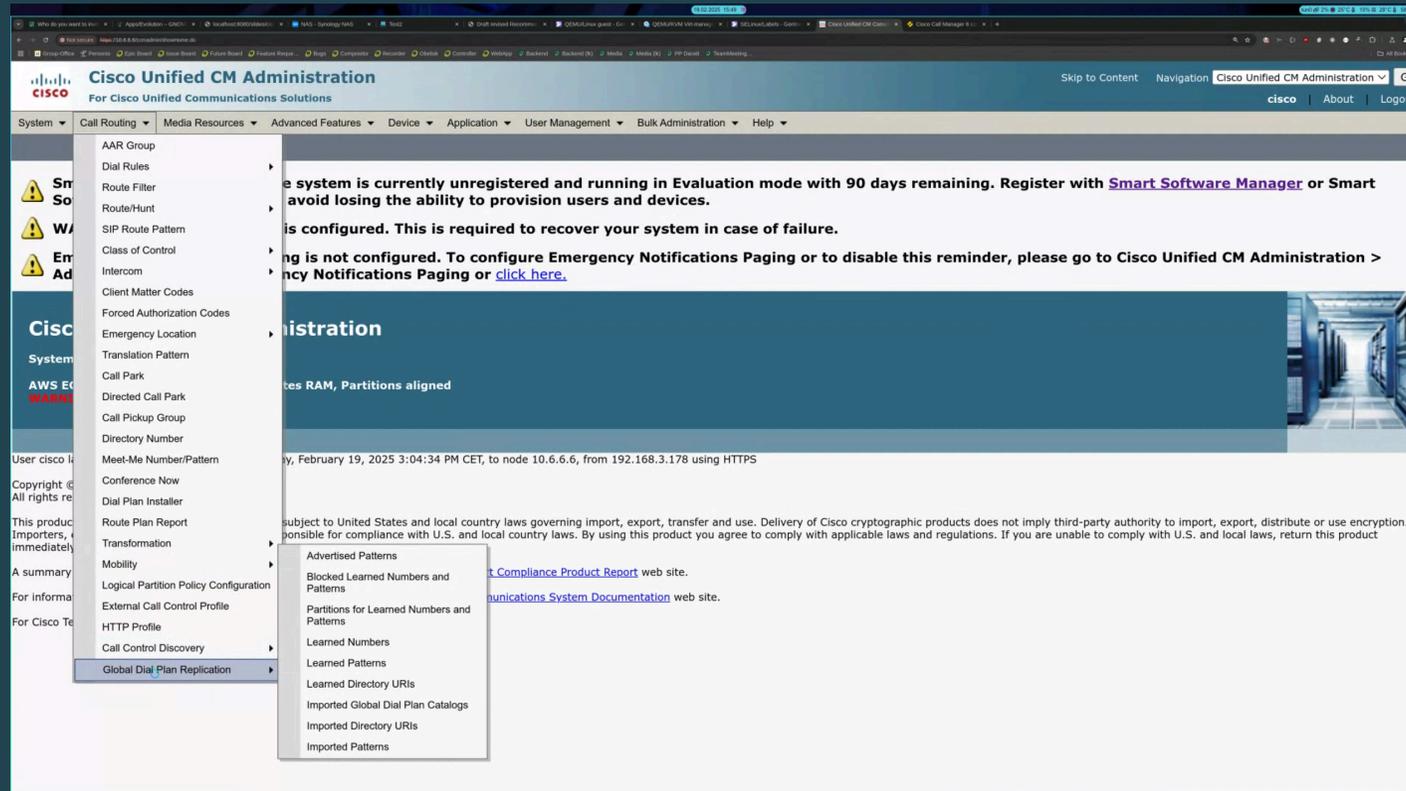


Egal, erstmal anmelden ...

The screenshot shows the Cisco Unified CM Administration web interface. At the top left is the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". To the right of the header are links for "Skip to Content", "Navigation", a dropdown menu for "Cisco Unified CM Administration", and a "Go" button. Below the header is a navigation bar with links for "System", "Call Routing", "Media Resources", "Advanced Features", and "Device". The main content area is mostly blank, with a central blue box containing the text "Loading, please wait."



Wie schlimm kann eine WebUI denn bitte sein?





Egal, erstmal Call-Service aktivieren ...

Call-Service aktivieren beim CallManager?



Erstmal die Call-Services aktivieren

Cisco Unified Serviceability
For Cisco Unified Communications Solutions

Navigation: Cisco Unified Serviceability Go

Alarm Trace Tools Snmp CallHome Help

Service Activation Related Links: Control Center - Feature Services Go

Save Set to Default Refresh

Status
Ready

Select Server
Server* cucm--CUCM Voice/Video Go
 Check All Services

CM Services

	Service Name	Activation Status
<input checked="" type="checkbox"/>	Cisco CallManager	Deactivated
<input type="checkbox"/>	Cisco Unified Mobile Voice Access Service	Deactivated
<input checked="" type="checkbox"/>	Cisco IP Voice Media Streaming App	Deactivated
<input type="checkbox"/>	Cisco CTIManager	Deactivated
<input type="checkbox"/>	Cisco Extension Mobility	Deactivated
<input type="checkbox"/>	Cisco Extended Functions	Deactivated
<input type="checkbox"/>	Cisco DHCP Monitor Service	Deactivated
<input type="checkbox"/>	Cisco Intercluster Lookup Service	Deactivated
<input type="checkbox"/>	Cisco Location Bandwidth Manager	Deactivated
<input type="checkbox"/>	Cisco Directory Number Alias Sync	Deactivated
<input type="checkbox"/>	Cisco Directory Number Alias Lookup	Deactivated
<input type="checkbox"/>	Cisco Headset Service	Deactivated
<input checked="" type="checkbox"/>	Cisco Device Activation Service	Activated
<input type="checkbox"/>	Cisco Dialed Number Analyzer Server	Deactivated
<input type="checkbox"/>	Cisco Dialed Number Analyzer	Deactivated
<input checked="" type="checkbox"/>	Cisco Tftp	Deactivated

CTI Services

	Service Name	Activation Status
<input type="checkbox"/>	Cisco IP Manager Assistant	Deactivated
<input type="checkbox"/>	Cisco WebDialer Web Service	Deactivated
<input type="checkbox"/>	Self Provisioning IVR	Deactivated



Erstmal die Call-Services aktivieren

Cisco Unified Serviceability
For Cisco Unified Communications Solutions

Navigation: Cisco Unified Serviceability Go

Alarm Trace Tools Snmp CallHome Help

Service Activation Related Links: Control Center - Feature Services Go

Save Set to Default Refresh

Status
Ready

Select Server
Server* cucm--CUCM Voice/Video Go
 Check All Services

CM Services

Service Name	Activation Status
<input checked="" type="checkbox"/> Cisco CallManager	Deactivated
<input type="checkbox"/> Cisco Unified Mobile Voice Access Service	Deactivated
<input checked="" type="checkbox"/> Cisco IP Voice Media Streaming App	Deactivated
<input type="checkbox"/> Cisco CTIManager	Deactivated
<input type="checkbox"/> Cisco Extension Mobility	Deactivated
<input type="checkbox"/> Cisco Extended Functions	Deactivated
<input type="checkbox"/> Cisco DHCP Monitor Service	Deactivated
<input type="checkbox"/> Cisco Intercluster Lookup Service	Deactivated
<input type="checkbox"/> Cisco Location Bandwidth Manager	Deactivated
<input type="checkbox"/> Cisco Directory Number Alias Sync	Deactivated
<input type="checkbox"/> Cisco Directory Number Alias Lookup	Deactivated
<input type="checkbox"/> Cisco Headset Service	Deactivated
<input checked="" type="checkbox"/> Cisco Device Activation Service	Activated
<input type="checkbox"/> Cisco Dialed Number Analyzer Server	Deactivated
<input type="checkbox"/> Cisco Dialed Number Analyzer	Deactivated
<input checked="" type="checkbox"/> Cisco Tftp	Deactivated

Loading, please wait.

CTI Services

Service Name	Activation Status
<input type="checkbox"/> Cisco IP Manager Assistant	Deactivated
<input type="checkbox"/> Cisco WebDialer Web Service	Deactivated
<input type="checkbox"/> Self Provisioning IVR	Deactivated



Schwamm drüber, noch 22 weitere Schritte



Date/Time Group Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Date/Time Group Configuration

Save Delete Copy Reset Add New

Status

Status: Ready

Date/Time Group Information

Date/Time Group: Zenitel (used by 3 devices)

Group Name*

Time Zone*

Separator* (applies to Date Format only)

Date Format*

Time Format*

Phone NTP References for this Date/Time Group

Selected Phone NTP References**

*- indicates required item.

**Selected Phone NTP References are ordered by highest priority

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Region Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Region Configuration

Related Links: [Back To Find/List](#)

Save Delete Reset Add New

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
NOTE: Region(s) not displayed	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Zenitel"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

*- indicates required item.

**The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



SIP Trunk Security Profile Configuration

The screenshot shows the 'SIP Trunk Security Profile Configuration' page. At the top, there are navigation tabs: System, Call Routing, Media Resources, Voice Mail, Device, Application, and User. Below the title bar, there are action buttons: Save, Delete (with a red X), Copy, Reset, and Add New. The 'Status' section shows 'Status: Ready'. The main configuration area is titled 'SIP Trunk Security Profile Information' and contains the following fields and options:

- Name*: AlphaCom
- Description: AlphaCom SIP Secure Profile
- Device Security Mode: Non Secure
- Incoming Transport Type*: TCP+UDP
- Outgoing Transport Type: UDP
- Enable Digest Authentication
- Nonce Validity Time (mins)*: 600
- X.509 Subject Name: (empty)
- Incoming Port*: 5060
- Enable Application Level Authorization
- Accept Presence Subscription
- Accept Out-of-Dialog REFER
- Accept Unsolicited Notification
- Accept Replaces Header

At the bottom, there are buttons for Save, Delete, Copy, Reset, and Add New. A note at the bottom left states: '*- indicates required item.'

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Partition Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾

Partition Configuration

Save Delete Reset Add New

Status

Status: Ready

Partition Information

Name*

Description

Time Schedule

Time Zone Originating Device
 Specific Time Zone

*- indicates required item.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Calling Search Space Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Manager ▾

Calling Search Space Configuration

Save Delete Copy Add New

Status

Status: Ready

Calling Search Space Information

Name*

Description

Route Partitions for this Calling Search Space

Available Partitions**

▼ ▲

Selected Partitions

▼ ▲

*- indicates required item.

**Selected Partitions are ordered by highest priority

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Media Resource Group Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Media Resource Group Configuration

Save **X** Delete Copy Reset + Add New

Status
Status: Ready

Media Resource Group Status
Media Resource Group: AlphaCom_MRG (used by 3 devices)

Media Resource Group Information

Name* AlphaCom_MRG
Description AlphaCom_MRG

Devices for this Group

Available Media Resources**
ANN_2
CFB_2
MOH_2

Selected Media Resources*
MTP_2 (MTP)

Use Multicast for MOH Audio (If at least one multicast MOH resource is available)

Save Delete Copy Reset Add New

i *- indicates required item.
i **Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Media Resource Group List Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ BU

Media Resource Group List Configuration

Save Delete Copy Reset Add New

Status
 Status: Ready

Media Resource Group List Status
Media Resource Group List: AlphaCom_MRGL (used by 3 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List
Available Media Resource Groups

▼ ▲

Selected Media Resource Groups

▼ ▲

*- indicates required item.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Device Pool Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ BU

Media Resource Group List Configuration

Save Delete Copy Reset Add New

Status
 Status: Ready

Media Resource Group List Status
Media Resource Group List: AlphaCom_MRGL (used by 3 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List
Available Media Resource Groups
Selected Media Resource Groups

*. indicates required item.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Device Pool Configuration, continued

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk ▾

Device Pool Configuration

Save Delete Copy Reset Add New

Status
Status: Ready

Device Pool Information
Device Pool: Zenitel (3 members**)

Device Pool Settings

Device Pool Name*	Zenitel
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	DefaultUser
Reverted Call Focus Priority	Default

Roaming Sensitive Settings

Date/Time Group*	Zenitel
Region*	Zenitel
Media Resource Group List	AlphaCom_MRGL
Location	Hub_None
Network Locale	< None >
SRST Reference*	Use Default Gateway
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Media Termination Point Configuration

Device Mobility Related Information****

Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >

Save Delete Copy Reset Add New

i *- indicates required item.

i **Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

i ***leave blank to use default.

i ****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Trunk Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾

Media Termination Point Configuration

Save Reset

Status

Status: Ready

Media Termination Point Information

Registration	Registered with Cisco Unified Communications Manager 10.5.2.43
IP Address	10.5.2.43
Media Termination Point Type*	Cisco Media Termination Point Software
Host Server*	10.5.2.43
Media Termination Point Name*	<input type="text" value="MTP_2"/>
Description	<input type="text" value="MTP_ccm"/>
Device Pool*	<input type="text" value="Zenitel"/>

*- indicates required item.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Trunk Configuration, continued

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User M

Trunk Configuration

Save ✖ Delete 🔄 Reset + Add New

Status
Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Device Name*	AlphaCom
Description	AlphaCom
Device Pool*	Zenitel
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	AlphaCom_MRGL
Location*	Hub_None
AAR Group	< None >
Packet Capture Mode*	None
Packet Capture Duration	0

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information
MLPP Domain < None >

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Trunk Configuration, continued

Call Routing Information	
Inbound Calls	
Significant Digits*	All
Connected Line ID Presentation*	Default
Connected Name Presentation*	Default
Calling Search Space	DefaultUser
AAR Calling Search Space	< None >
Prefix DN	
<input type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	
Outbound Calls	
Calling Party Selection*	Originator
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Caller ID DN	
Caller Name	
<input type="checkbox"/> Redirecting Diversion Header Delivery - Outbound	

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Route Group Configuration

SIP Information	
Destination Address*	<input type="text" value="10.5.101.80"/>
<input type="checkbox"/> Destination Address is an SRV	
Destination Port*	<input type="text" value="5060"/>
MTP Preferred Originating Codec*	<input type="text" value="711ulaw"/>
Presence Group*	<input type="text" value="Standard Presence group"/>
SIP Trunk Security Profile*	<input type="text" value="AlphaCom"/>
Rerouting Calling Search Space	<input type="text" value=" < None >"/>
Out-Of-Dialog Refer Calling Search Space	<input type="text" value=" < None >"/>
SUBSCRIBE Calling Search Space	<input type="text" value=" < None >"/>
SIP Profile*	<input type="text" value="Standard SIP Profile"/>
DTMF Signaling Method*	<input type="text" value="No Preference"/>

i *- indicates required item.

i **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Route List Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Group Configuration

Save Delete Add New

Route Group Information

Route Group Name*
Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains

Available Devices**

Port(s)

Current Route Group Members

Selected Devices***

Removed Devices****

Route Group Members

AlphaCom

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Route Pattern Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾

Route Pattern Configuration

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Route Pattern Configuration, continued

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

*- indicates required item.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Phone Configuration

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Phone Configuration

Save ✖ Delete 📄 Copy 🔄 Reset + Add New

Status
Status: Ready

Association Information
Modify Button Items

1	Line [1] - 1030 in AllLines
2	Line [2] - Add a new DN
3	Add a new SD
4	Add a new SD
5	Add a new SD
6	Add a new SD
----- Unassigned Associated Items -----	
7	Add a new SD
8	Add a new SURL
9	Add a new BLF SD
10	Add a new BLF Directed Call Park
11	Privacy
12	None

Phone Type
Product Type: Cisco 7960
Device Protocol: SCCP

Device Information

Registration	Registered with Cisco Unified Communications Manager 10.5.2.43
IP Address	10.5.101.122
MAC Address*	0006D74B1DA0
Description	Zenitel 1030
Device Pool*	Zenitel View Details
Common Device Configuration	< None > View Details
Phone Button Template*	Standard 7960 SCCP
Softkey Template	< None >
Common Phone Profile*	Standard Common Phone Profile
Calling Search Space	< None >
AAR Calling Search Space	< None >
Media Resource Group List	< None >
User Hold MOH Audio Source	< None >
Network Hold MOH Audio Source	< None >
Location*	Hub_None
AAR Group	< None >
User Locale	< None >
Network Locale	< None >

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Phone Configuration, continued

Built In Bridge*	Default	▼
Privacy*	Default	▼
Device Mobility Mode*	Default	▼
Owner User ID	< None >	▼
Phone Load Name		
Join Across Lines	Default	▼
BLF Audible Alert Setting (Phone Idle)*	Default	▼
BLF Audible Alert Setting (Phone Busy)*	Default	▼
<input checked="" type="checkbox"/> Is Active		
<input checked="" type="checkbox"/> Retry Video Call as Audio		
<input type="checkbox"/> Ignore Presentation Indicators (internal calls only)		
<input checked="" type="checkbox"/> Allow Control of Device from CTI		
<input checked="" type="checkbox"/> Logged Into Hunt Group		
<input type="checkbox"/> Remote Device		
Protocol Specific Information		
Packet Capture Mode*	None	▼
Packet Capture Duration	0	
Presence Group*	Standard Presence group	▼
Device Security Profile*	Cisco 7960 - Standard SCCP Non-Secure Profile	▼
SUBSCRIBE Calling Search Space	< None >	▼
<input type="checkbox"/> Unattended Port		
<input type="checkbox"/> Require DTMF Reception		
<input type="checkbox"/> RFC2833 Disabled		

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Phone Configuration, continued

Certification Authority Proxy Function (CAPF) Information	
Certificate Operation*	<input type="text" value="No Pending Operation"/>
Authentication Mode*	<input type="text" value="By Null String"/>
Authentication String	<input type="text"/>
<input type="button" value="Generate String"/>	
Key Size (Bits)*	<input type="text" value="1024"/>
Operation Completes By	<input type="text" value="2009"/> <input type="text" value="1"/> <input type="text" value="25"/> <input type="text" value="12"/> (YYYY:MM:DD:HH)
Certificate Operation Status: None	
Note: Security Profile Contains Addition CAPF Settings.	

Expansion Module Information	
Module 1	<input type="text" value="< None >"/>
Module 1 Load Name	<input type="text"/>
Module 2	<input type="text" value="< None >"/>
Module 2 Load Name	<input type="text"/>

External Data Locations Information (Leave blank to use default)	
Information	<input type="text"/>
Directory	<input type="text"/>
Messages	<input type="text"/>
Services	<input type="text"/>
Authentication Server	<input type="text"/>
Proxy Server	<input type="text"/>
Idle	<input type="text"/>
Idle Timer (seconds)	<input type="text"/>

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Phone Configuration, continued

Extension Information	
<input type="checkbox"/> Enable Extension Mobility	
Log Out Profile	-- Use Current Device Settings --
Log in Time	< None >
Log out Time	< None >

MLPP Information	
MLPP Domain	< None >
MLPP Indication*	Default
MLPP Preemption*	Default

Do Not Disturb	
<input type="checkbox"/> Do Not Disturb	
DND Option*	Ringer Off
DND Incoming Call Alert	< None >

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Phone Configuration, continued

Product Specific Configuration Layout 

Disable Speakerphone

Disable Speakerphone and Headset

PC Port *

Settings Access*

Gratuitous ARP*

PC Voice VLAN Access*

Video Capabilities*

Auto Line Select*

Web Access*

 *- indicates required item.

 **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

 ***Note: Security Profile Contains Addition CAPF Settings.

Quelle: https://wiki.zenitel.com/wiki/Cisco_Call_Manager_6_configuration_guide



Please stop ...



Quelle: <https://www.kapwing.com/explore/crying-cat-meme-template>



Setup Cisco 8845





Setup Cisco 8845 (800% Zoom)

14 Answer Oldest

15 [Add a new BLF Directed Call Park](#)

16 Call Park

17 Call Pickup

18 CallBack

19 Do Not Disturb

20 Group Call Pickup

21 Hunt Group Logout

22 [Intercom \[1\] - Add a new Intercom](#)

23 Malicious Call Identification

24 Meet Me Conference

25 Mobility

26 Other Pickup

27 Quality Reporting Tool

28 Queue Status

29 Redial

30 [Add a new SURF](#)

31 Services

32 [Add a new BLF SD](#)

33 Privacy

34 None

Calling Search Space < None >

AAR Calling Search Space < None >

Media Resource Group List < None >

User Hold MOH Audio Source < None >

Network Hold MOH Audio Source < None >

Location * Hub_None

AAR Group < None >

User Locale < None >

Network Locale < None >

Built In Bridge * Default

Privacy * Default

Device Mobility Mode * Default [View Current Device Mobility Settings](#)

Owner User Anonymous (Public/Shared Space)

Owner User ID * 1000

Mobility User ID < None >

Phone Personalization * Default

Services Provisioning * Default

Phone Load Name

Use Trusted Relay Point * Default

BLF Audible Alert Setting (Phone Idle) * Default

BLF Audible Alert Setting (Phone Busy) * Default

Always Use Prime Line * Default

Always Use Prime Line for Voice Message * Default

Geolocation

Ignore Presentation Indicators (internal calls only)

Allow Control of Device from CTI

Logged Into Hunt Group

Remote Device

Protected Device ****

Hot line Device *****

Require off-premise location

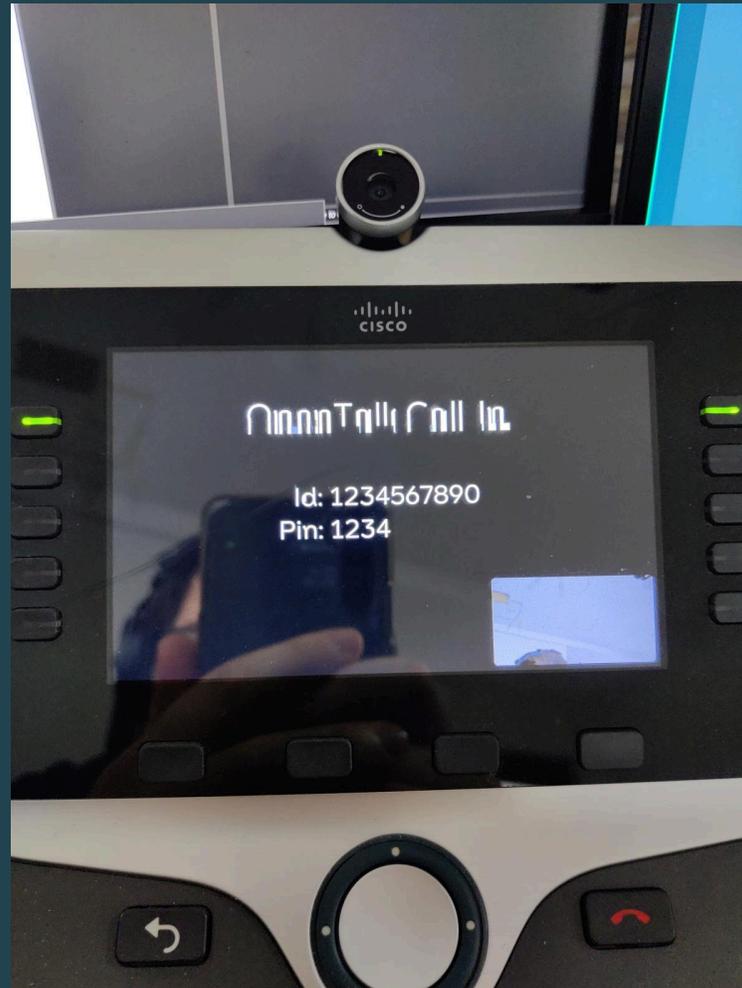


Genug Einrichtung!

Demo mit CUCM (Cisco 8945).



Warte ... das kennen wir doch?!





Schauen wir uns den SDP-Invite nochmal an

```
m=video 14184 RTP/AVP 103 99\r\n\r\na=crypto:1 AES_CM_128_HMAC_SHA1_80 inline:CNI3haHVYr2rIME59idfVnkL+rRm70kldGIx1r00\r\n\r\na=ice-ufrag:3cc7bc704d150b5f4a71ed0b73ed5dcb\r\na=ice-pwd:7626025f51a15160720372474984ccccf\r\na=candidate:Hc0a803b2 1 UDP 2130706431 192.168.3.178 14184 typ host\r\na=candidate:Hac110001 1 UDP 2130706431 172.17.0.1 14184 typ host\r\n\r\n...\r\na=candidate:Hc0a87a01 2 UDP 2130706430 192.168.122.1 14185 typ host\r\na=candidate:Ha003016 2 UDP 2130706430 10.0.48.22 14185 typ host\r\n\r\na=rtpmap:103 h263-1998/90000\r\na=fmtp:103 CIF4=1;CIF=1;QCIF=1;CUSTOM=352,240,1;MaxBR=30000\r\n\r\na=rtpmap:99 H264/90000\r\na=fmtp:99 max-mbps=245000;max-fs=8160;max-br=2500;max-smbps=245000;max-fps=3000;packetization-mode=0;profile-level-id=428014\r\n\r\na=sendrecv\r\n
```



SDP-Invite (nur Video-Bereich)

```
m=video 14184 RTP/AVP 103 99\r\n  
a=rtpmap:99 H264/90000\r\n  
a=fmtp:99 max-mps=245000;max-fs=8160;max-br=2500;max-smps=245000;max-fps=3000;packetization-mode=0;profile-level-id=428014\r\n  
a=sendrecv\r\n
```



Die Lösung

Bitrate richtig berechnen!

Encoder richtig einstellen!



Genug Einrichtung!

Demo mit CUCM (Webex DX80).



Warte ... das kennen wir doch?!



Hatten wir die Bitrate nicht gefixed?



Die Lösung

Bitrate durch Zwei teilen!

- Cisco geht von "Duplex" aus (Hin- und Rück-Transport)

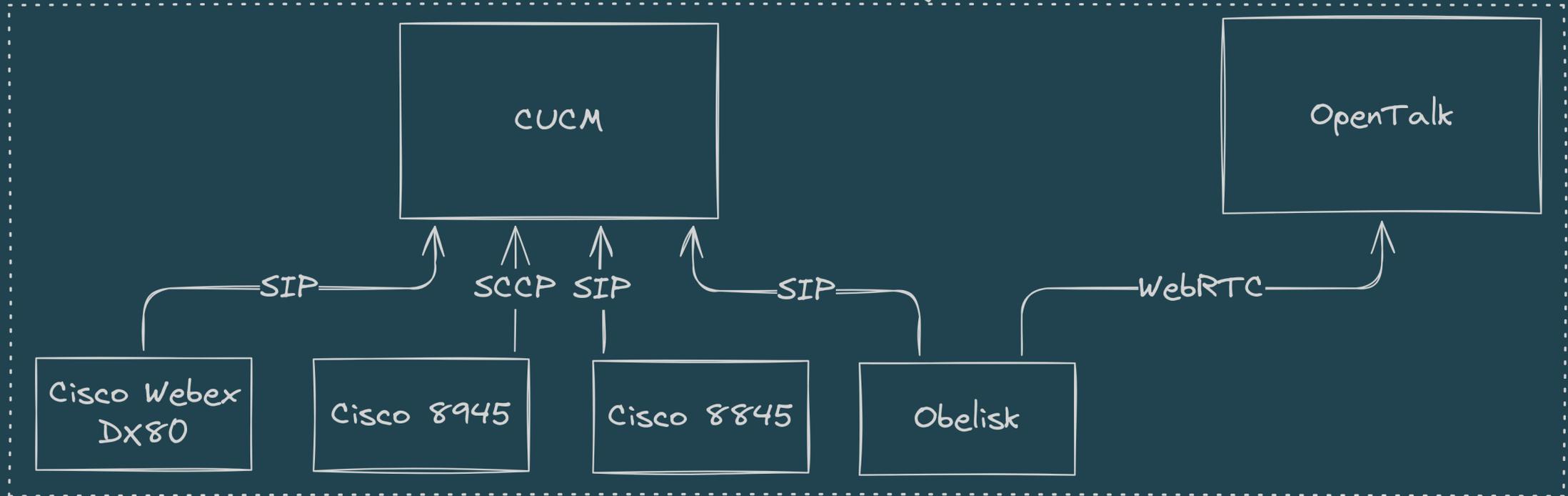


Kleinere Fuckups



"Fun" mit dem Cisco 8945

Kommunikation mittels Cisco Unified Communications Manager (CUCM)



Senden der Kamera -> Crash des Telefons (Obelisk)



"Fun" mit Langzeittests

- Verbindung hängen geblieben nach 15 Minuten
- Decoder-Error nach 15 Minuten (Video bleibt stehen)
- Lösung: Senden bei Decoder-Fehlern ein Keyframe-Request in XML über SIP



Scheinbar kein ICE-Support

- Standardmäßig nicht an
- Keine Doku gefunden
- Nach Consultig-Anfrage angeblich nicht vorhanden
- Netzwerk-Topologien zu überkreuzen (sprich verschiedene Netzwerke)



SRTP (Secure RTP)

- Ist eine eigene Lizenz
- "Kostenlos" bei jeder Telefon-Lizenz dabei
- Verwirrend aufzusetzen (kein Guide)



Lizenzmodell

- Intransparent
- Verwirrend

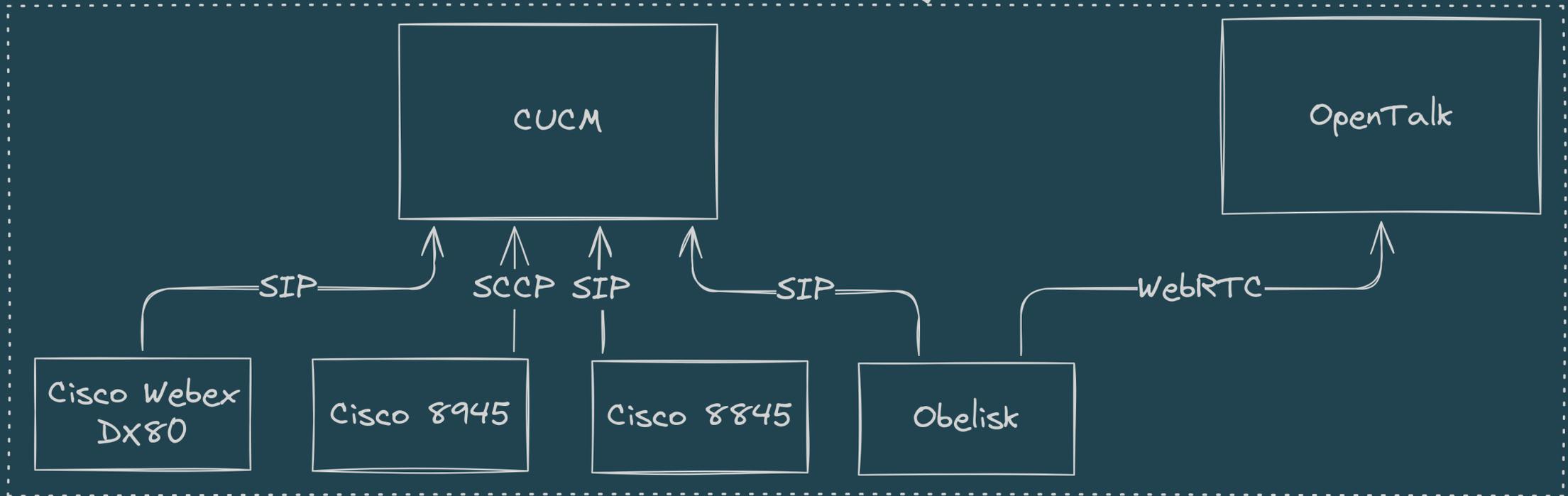


Vendor-Lock

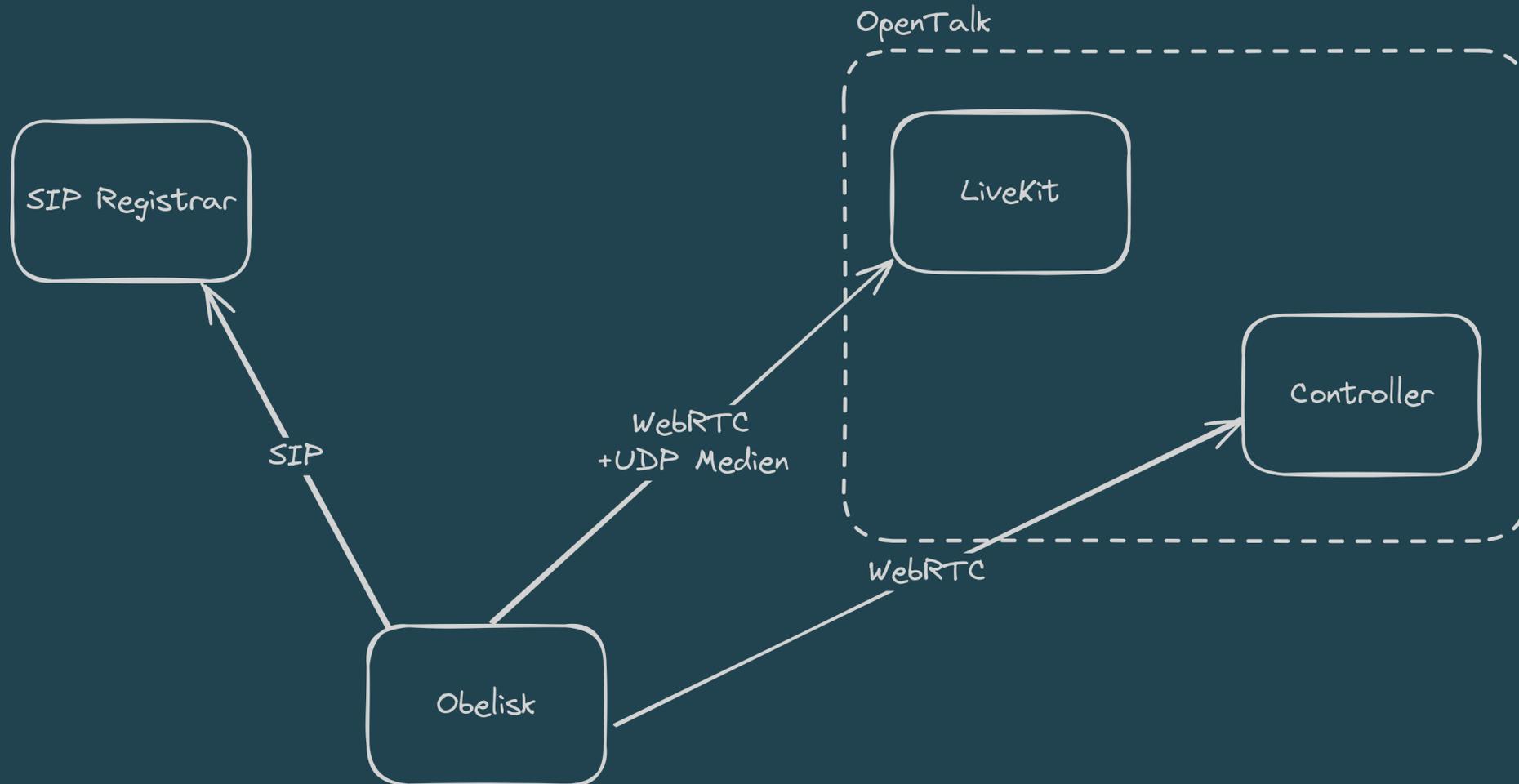
- Webex DX80 möchte "nach Hause telefonieren"
- Upgrader der Firmware unmöglich
- Startup-Netzwerk-Check
- Reset hat nichts gebracht



Kommunikation mittels Cisco Unified Communications Manager (CUCM)



LiveKit





Wechsel von Janus zu LiveKit

- LiveKit skaliert extrem gut
- Fertige Helm-Charts
- Rust SDK
- Stabilere Verbindung



We appreciate Rust

- Komplette Codebase ist in Rust
- Alle Services (bis auf das Frontend)
- Kein GStreamer
- Alles selbst in Rust und mit SIMD neu geschrieben
- Kudos an das Team



Warum der ganze Schmerz?

- Weil wir es können ... ;)



Warum der ganze Schmerz?

- Kunden haben viele Cisco Geräte
- Geräte wurden teuer angeschafft
- Geräte vom Aussterben retten (Nachhaltigkeit, reduce, reuse, ...)



Outcome

- SIP ist nicht gleich SIP
- Standard ist schwammig und jeder Hersteller hat seine eigene Auslegung
- Eigene Auslegung selbst Modellzpezifisch (in der gleichen Firma)

Dankeschön!



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*Alle Diagramme wurden mit Excalidraw gezeichnet