

# PROTOCOL

to exercise

**VoIP**

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**VoIP**

## Used Programs

Nr.	Device	Manufacturer	Type	Version
1.	Phoner / Telephone (Mac)	-	Tool	

## 1 Task

The task of this laboratory exercise was to build a connection with voice over IP. Wireshark was to be used to see what the program Phoner does in the background, while connecting

## 2 Setting up Phoner

Firstly, Phoner has to be set up right in order to be able to connect to the SIP-Proxy (Compact 5020). The IP of the Auerswald device is 10.81.229.70, so this IP has been set as the proxy/register. As the preferred kind of connection, UDP has been chosen. Because of the settings of the Compact 5020, a telephone number between 70 and 73 had to be chosen, for this exercise, 71 was the choice. For the Realm, the same IP as the one for proxy/register was the right one. The local Port was preconfigured as 5060.

The screenshot shows the 'Geräte' (Devices) configuration window in Phoner. The 'SIP' tab is selected. The settings are as follows:

- Profil:** 10.81.229.70 (with a 'Löschen' button)
- Benutzername:** 71
- Passwort:** (empty)
- Angezeigter Name:** Hirsch
- Proxy/Registrar:** 10.81.229.70 (with a checked 'Registrierung' checkbox)
- STUN Server:** (empty)
- Lokaler Port:** 5060 (with a spinner)
- bevorzugte Verbindungsart:** UDP (selected), TCP, TLS
- Realm:** 10.81.229.70
- Telefonnummer:** 71
- API:** SIP (selected), TAPI, CAPI
- Additional options:** ☒ CAPI zusätzlich aktivieren, ☐ ZRTP, ☐ UPnP NAT, ☐ IPv6
- Buttons:** 'Codecs' (with a gear icon), 'OK', 'Abbruch'
- Stille-Erkennung:** keine Stille-Erkennung (dropdown)

Now the same settings have to be set on a second device:

Geräte

TAPI CAPI **SIP**

Profil  
 Settings Löschen

Benutzername: 70 Passwort:  Angezeigter Name: Biehl

Proxy/Registrar: 10.81.229.70 ☒ Registrierung STUN Server:

Lokaler Port: 5060  bevorzugte Verbindungsart:  
☒ UDP ☐ TCP ☐ TLS

☒ CAPI zusätzlich aktivieren ☐ ZRTP ☐ UPnP NAT ☐ IPv6 Codecs

Realm: 10.81.229.70 Telefonnummer: 70

keine Stille-Erkennung

API  
☐ TAPI ☐ CAPI ☒ SIP

OK Abbruch

### 3 Connecting to device

While trying to connect to another VOIP device, Phoner tries that by using following protocols

29	64.93983200(10.81.229.70	10.211.55.3	SIP	400 Status: 100 Trying
30	65.39929500(10.81.229.70	10.211.55.3	SIP/SDP	968 Status: 183 Session Progress
47	65.59371700(10.81.229.70	10.211.55.3	SIP	613 Status: 180 Ringing

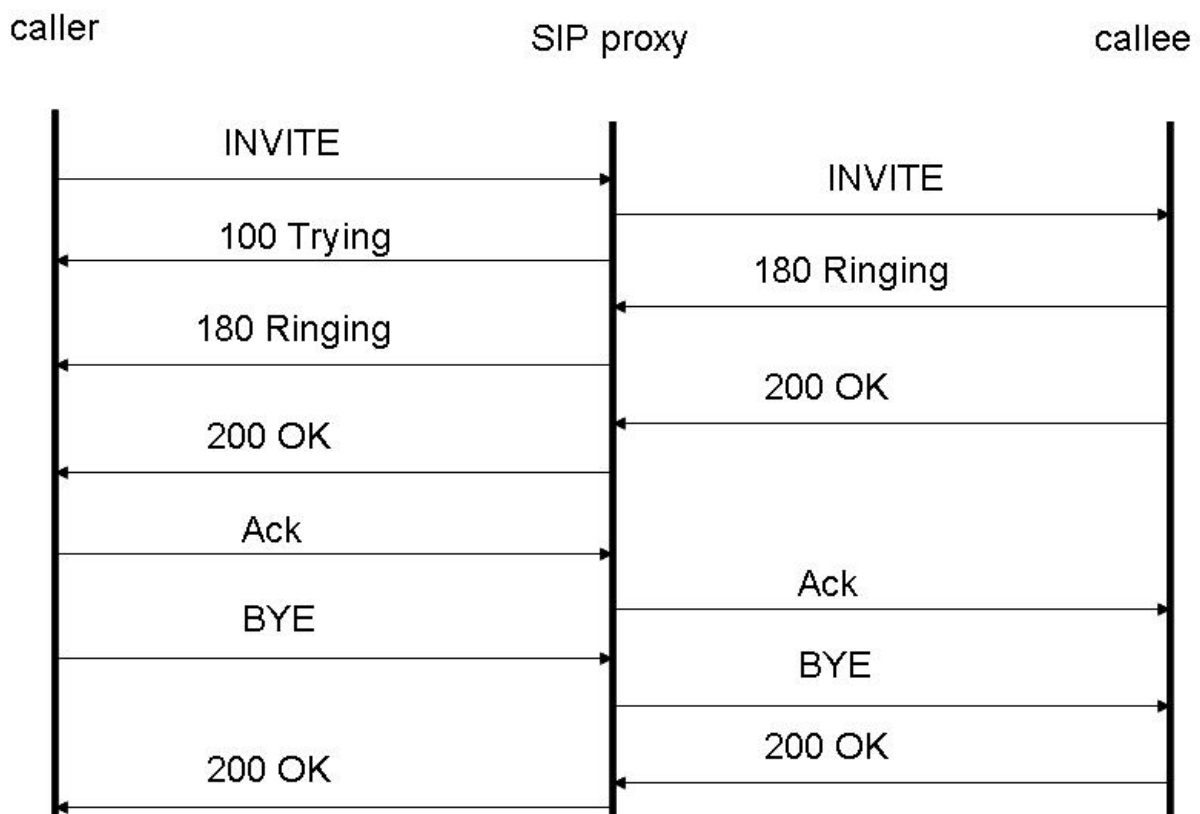
Wirshark showed, that the standard Protocol for VOIP, SIP was being used.

## 4 Calling

If somebody calls you, a small message shows up at the screen. The name and the number are shown.



The call is always in the same format. The following diagram shows the process not only during also before and after the call. The structure is the same as in this case.



To prove that the construction corresponds the truth some screenshots from wireshark have been made.

Wireshark reports all packages which are sent.

Call try

273151	2972.67318	10.85.248.118	10.85.249.99	SIP	433 Status: 100 Trying
273152	2972.68537	10.85.248.118	10.85.249.99	SIP	503 Status: 180 Ringing
273164	2974.83275	10.85.248.118	10.85.249.99	SIP/SDF	1043 Status: 200 OK

Shortly after the call is trying it goes on ringing

### Ringing

273171	2974.85174	10.85.248.118	10.85.249.99	RTP	214 PT=ITU-T G.711 PCMA, SSRC=0x624860D0, Seq=25758, Time=160, Mark
273172	2974.86978	10.85.249.99	10.85.248.118	RTP	214 PT=ITU-T G.711 PCMA, SSRC=0x3C564132, Seq=25196, Time=2480
273174	2974.87156	10.85.248.118	10.85.249.99	RTP	214 PT=ITU-T G.711 PCMA, SSRC=0x624860D0, Seq=25759, Time=320
273175	2974.89043	10.85.249.99	10.85.248.118	RTP	214 PT=ITU-T G.711 PCMA, SSRC=0x3C564132, Seq=25197, Time=2640
273176	2974.89158	10.85.248.118	10.85.249.99	RTP	214 PT=ITU-T G.711 PCMA, SSRC=0x624860D0, Seq=25760, Time=480

During the Ringing you can see some Packages with the timelap

### Call denied

290722	3011.32863	10.85.249.99	10.85.250.62	SIP	433 Status: 100 Trying
290725	3011.33751	10.85.249.99	10.85.250.62	SIP	503 Status: 180 Ringing
290842	3012.32109	10.85.250.62	10.85.249.99	SIP	466 Request: CANCEL sip:70810.81.229.70

After the ringing the call was cancelled

### Getting a call

307404	3392.84694	10.85.250.62	10.85.249.99	SIP/SDF	1098 Request: INVITE sip:70810.81.229.70
307405	3392.84866	10.85.249.99	10.85.250.62	SIP	433 Status: 100 Trying
307406	3392.85955	10.85.249.99	10.85.250.62	SIP	503 status: 180 Ringing
307437	3396.05904	10.85.249.99	10.85.250.62	SIP/SDF	1039 Status: 200 OK
307439	3396.06521	10.85.250.62	10.85.249.99	RTCP	98 Sender Report Source description
307440	3396.06538	10.85.250.62	10.85.249.99	RTP	214 PT=ITU-T G.711 PCMA, SSRC=0x43D6C6BC, Seq=40808, Time=3370, Mark
307441	3396.06817	10.85.249.99	10.85.250.62	RTCP	98 Sender Report Source description

If you get a call the beginning looks the same but as soon as you accept it the packages change a little bit.

### During a call

294015	3074.18712	10.85.250.63	10.85.249.99	UDP	1274 Source port: ws-discovery Destination port: 58752
294089	3074.28646	10.81.230.21	10.85.249.99	UDP	1270 Source port: ws-discovery Destination port: 58752
294138	3074.30404	10.85.251.62	10.85.249.99	UDP	1267 Source port: ws-discovery Destination port: 58752

The normal packages over the right port can be seen.

### Close the call

307757	3398.80278	10.85.249.99	10.85.250.62	SIP	462 Request: BYE sip:73@10.85.250.62:5060
307758	3398.80461	10.85.250.62	10.85.249.99	SIP	428 Status: 200 OK
307761	3399.11443	10.85.250.62	10.85.249.99	RTCP	122 Sender Report Source description

The call is going to be interrupted and then it ends