

## Konfiguration des Grandstream HT 286 zur Anbindung der SIEDLE DCA 612-0



### Konfiguration des Asterisk-Servers

Diese Dokumentation beschreibt die Anbindung der Wechselsprechanlage SIEDLE DCA 612-0 an den Asterisk-Server. Die DCA 612-0 verfügt über einen analogen Ausgang (Western Stecker) und kann darüber an einen Analog-IP-Konverter angeschlossen werden. Zum Einsatz kam in diesem Projekt ein Grandstream HT 286, da dieser den erforderlichen Port bietet und relativ preiswert ist.

Auf dem Asterisk Server ist für den HT 286 ein SIP-Account anzulegen, damit er sich am Asterisk-Server anmelden kann:

#### **`/etc/asterisk/sip.conf`**

```
[Tuer]
type=friend
callerid="Tuer" <181>
host=dynamic
username=Tuer
secret=Tuer181
context=tuer
```

Der in der sip.conf definierte Kontext tuer muß auch in der extensions.conf aufgeführt sein:

#### **`/etc/asterisk/extensions.conf`**

```
[tuer]
    exten => s,1,DIAL(SIP/Zentrale,30,tr)
    exten => s,2,HANGUP()
```

Klingelt es an der Tür, kann die Verbindung zur DCA über eine der Tasten 0-9 hergestellt werden. Das Öffnen der Tür erfolgt durch die Eingabe von #61, nachdem man zuvor die Verbindung mit einer der Tasten 0-9 hergestellt hat.

Die Konfiguration des HT 286 kann den Bildschirmfotos auf den folgenden Seiten entnommen werden.

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### Konfiguration des Grandstream HT 286

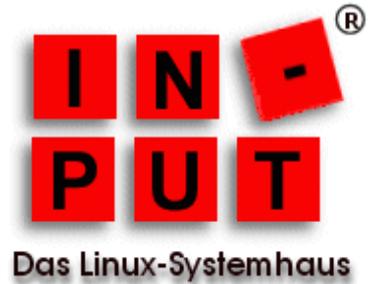
<b>Web Port:</b>	<input type="text" value="80"/> (default for HTTP is 80)
<b>IP Address:</b>	<input checked="" type="radio"/> dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)
DHCP hostname:	<input type="text"/>
DHCP domain:	<input type="text"/>
DHCP vendor class ID:	<input type="text"/>
PPPoE account ID:	<input type="text"/>
PPPoE password:	<input type="text"/>
Preferred DNS server:	<input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>
	<input checked="" type="radio"/> statically configured as:
IP Address:	<input type="text" value="192"/> <input type="text" value="168"/> <input type="text" value="0"/> <input type="text" value="160"/>
Subnet Mask:	<input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>
Default Router:	<input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>
DNS Server 1:	<input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>
DNS Server 2:	<input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/> <input type="text" value="0"/>
<b>Time Zone:</b>	<input type="text" value="GMT+1:00 (Paris, Amsterdam, Berlin, Rome, Vienna, Madrid, Warsaw, Brussels)"/> ▾
<b>Daylight Savings Time:</b>	<input checked="" type="radio"/> No <input type="radio"/> Yes
	Optional Rule: <input type="text" value="4,1,7,2,0;10,-1,7,2,0;60"/>

Ist im lokalen Netzwerk kein DHCP-Server im Einsatz, müssen die IP-Adresse, die Subnetz-Maske, der Default Router und mindestens ein DNS gesetzt werden.

<b>Admin Password:</b>	<input type="text"/>	(purposely not displayed for security protection)
<b>SIP Server:</b>	<input type="text" value="172.16.0.2"/>	(e.g., sip.mycompany.com, or IP address)
<b>Outbound Proxy:</b>	<input type="text"/>	(e.g., proxy.myprovider.com, or IP address, if any)
<b>SIP User ID:</b>	<input type="text" value="Tuer"/>	(the user part of an SIP address)
<b>Authenticate ID:</b>	<input type="text" value="Tuer"/>	(can be identical to or different from SIP User ID)
<b>Authenticate Password:</b>	<input type="text"/>	(purposely not displayed for security protection)
<b>Name:</b>	<input type="text" value="Tuer"/>	(optional, e.g., John Doe)
<b>Home NPA:</b>	<input type="text"/>	
<b>Advanced Options:</b>		
<b>Preferred Vocoder:</b> (in listed order)	choice 1:	<input pcmu""="" type="text" value="current setting is "/>
	choice 2:	<input pcma""="" type="text" value="current setting is "/>
	choice 3:	<input g723""="" type="text" value="current setting is "/>
	choice 4:	<input g729""="" type="text" value="current setting is "/>
	choice 5:	<input g726-32""="" type="text" value="current setting is "/>
	choice 6:	<input ilbc""="" type="text" value="current setting is "/>
	choice 7:	<input pcmu""="" type="text" value="current setting is "/>
	<b>G723 rate:</b>	<input checked="" type="radio"/> 6.3kbps encoding rate <input type="radio"/> 5.3kbps encoding rate

Die Adresse des SIP-Servers muß entsprechend angepaßt werden.

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*iLBC frame size:*  20ms  30ms  
*iLBC payload type:*  (between 96 and 127, default is 97)  
*Silence Suppression:*  No  Yes  
*Voice Frames per TX:*  (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)  
*Fax Mode:*  T.38 (Auto Detect)  Pass-Through  
*Layer 3 QoS:*  (Diff-Serv or Precedence value)  
*Layer 2 QoS:* 802.1Q/VLAN Tag  802.1p priority value  (0-7)  
*Allow incoming SIP messages from SIP proxy only:*  No  Yes  
*Use DNS SRV:*  No  Yes  
*User ID is phone number:*  No  Yes  
*SIP Registration:*  Yes  No  
*Unregister On Reboot:*  Yes  No  
*Register Expiration:*  (in seconds, default 1 hour, max 45 days)  
*Early Dial:*  No  Yes (use "Yes" only if proxy supports 484 response)  
*Allow outgoing call without Registration:*  No  Yes  
*Dial Plan Prefix:*  (this prefix string is added to each dialed number)  
*No Key Entry Timeout:*  (in seconds, default is 4 seconds)  
*Use # as Dial Key:*  No  Yes (if set to Yes, "#" will function as the Dial key)

*local SIP port:*  (default 5060)  
*local RTP port:*  (1024-65535, default 5004)  
*Use random port:*  No  Yes  
*NAT Traversal:*  No  
 Yes, STUN server is:  (URI or IP:port)  
*keep-alive interval:*  (in seconds, default 20 seconds)  
*Use NAT IP:*  (used in SIP/SDP message if specified)  
*Proxy-Require:*   
*SUBSCRIBE for MWI:*  No, do not send SUBSCRIBE for Message Waiting Indication  
 Yes, send periodical SUBSCRIBE for Message Waiting Indication  
*Offhook Auto-Dial:*  (User ID/extension to dial automatically when offhook)  
*Enable Call Features:*  No  Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)  
*Use Bell-style 3-way Conference:*  No  Yes (if Yes, \*23 will be disabled)  
*Disable Call-Waiting:*  No  Yes  
*Send DTMF:*  in-audio  via RTP (RFC2833)  via SIP INFO  
*DTMF Payload Type:*   
*Send Flash Event:*  No  Yes (Flash will be sent as a DTMF event if set to Yes)  
*Onhook Threshold:*

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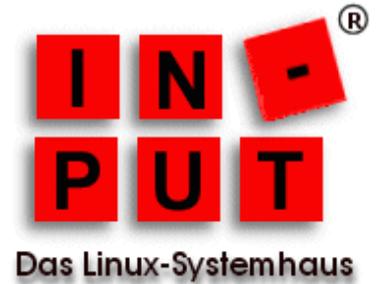


<i>FXS Impedance:</i>	CTR21 (270 Ohm + 750 Ohm  150nF)	▼
<i>Caller ID Scheme:</i>	ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA) ▼	
<i>Onhook Voltage:</i>	36V ▼	
<i>Polarity Reversal:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (reverse polarity upon call establishment and termination)	
<i>NTP Server:</i>	ptbtime1.ptb.de (URI or IP address)	
<i>Send Anonymous:</i>	<input checked="" type="radio"/> No <input type="radio"/> Yes (caller ID will be blocked if set to Yes)	
<i>Anonymous Method:</i>	<input checked="" type="radio"/> Use From Header <input type="radio"/> Use Privacy Header	
<i>Time to ring:</i>	60 seconds ▼	
<i>Special Feature:</i>	Standard ▼	
<i>Syslog Server:</i>		
<i>Syslog Level:</i>	NONE ▼	
<i>Session Expiration:</i>	180 (in seconds, default 180 seconds)	
<i>Min-SE:</i>	90 (in seconds, default and minimum 90 seconds)	
<i>Caller Request Timer:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No (Request for timer when making outbound calls)	
<i>Callee Request Timer:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No (When caller supports timer but did not request one)	
<i>Force Timer:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No (Use timer even when remote party does not support)	
<i>UAC Specify Refresher:</i>	<input type="radio"/> UAC <input type="radio"/> UAS <input checked="" type="radio"/> Omit (Recommended)	
<i>UAS Specify Refresher:</i>	<input checked="" type="radio"/> UAC <input type="radio"/> UAS (When UAC did not specify refresher tag)	
<i>Force INVITE:</i>	<input type="radio"/> Yes <input checked="" type="radio"/> No (Always refresh with INVITE instead of UPDATE)	
<i>Firmware Upgrade and</i>	Upgrade Via <input type="radio"/> TFTP <input checked="" type="radio"/> HTTP	

Betreibt man einen lokalen Zeitserver, kann statt der ptbtime1.ptb.de auch dieser gesetzt werden.

<i>Provisioning:</i>	Firmware Server Path: fm.grandstream.com/gs			
	Config Server Path: fm.grandstream.com/gs			
	Firmware File Prefix: <input type="text"/>	Firmware File Postfix: <input type="text"/>		
	Config File Prefix: <input type="text"/>	Config File Postfix: <input type="text"/>		
Automatic Upgrade:				
	<input type="radio"/> No <input checked="" type="radio"/> Yes, check for upgrade every 10080 minutes (default 7 days)			
	<input checked="" type="radio"/> Always Check for New Firmware			
	<input type="radio"/> Check New Firmware only when F/W pre/suffix changes			
	<input type="radio"/> Always Skip the Firmware Check			
<i>Firmware Key:</i>	<input type="text"/> (in Hexadecimal Representation)			
<i>Authenticate Conf File:</i>	<input type="radio"/> No <input checked="" type="radio"/> Yes (cfg file would be authenticated before acceptance if set to Yes)			
<i>Lock keypad update:</i>	<input type="radio"/> No <input checked="" type="radio"/> Yes (configuration update via keypad is disabled if set to Yes)			
<i>Allow conf SIP Account in Basic Settings:</i>	<input type="radio"/> No <input checked="" type="radio"/> Yes			
<i>Override MTU Size:</i>	<input type="text"/>			
<i>Volume Amplification:</i>	TX <input type="text"/> dB default RX <input type="text"/> dB default			
<i>Call Progress Tones:</i>	Frequency 1 (Hz)	Frequency 2 (Hz)	ON (x10ms) (C1;C2;C3)	OFF (x10ms) (C1;C2;C3)
Dial Tone	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>

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*Lock keypad update:*  No  Yes (configuration update via keypad is disabled if set to Yes)

*Allow conf SIP Account in Basic Settings:*  No  Yes

*Override MTU Size:*

*Volume Amplification:* TX  RX

*Call Progress Tones:*

	Frequency 1 (Hz)	Frequency 2 (Hz)	ON (x10ms) (C1;C2;C3)	OFF (x10ms) (C1;C2;C3)
Dial Tone	350	440	0	0
Recall Dial Tone	350	440	10	10
Message Waiting	350	440	10	10
Confirmation	350	440	10	10
Audible Ringing	440	480	200	400
Busy Tone	480	620	50	50
Reorder Tone	480	620	25	25
Receiver Offhook Tone	1400	2600	10	10

### Links

Grandstream HT 286: <http://grandstream.com/ht286.html>

SIEDLE: <http://www.siedle.de/>