

Peoplefone Trunk

Routes - SIP gateways

Title:	Peoplefone Deutschland	sip.conf:
Name:	gw_2_peoplefonedeuts	[peer-name]
Registrar / Server:	sip.peoplefone.de	host
Proxy [1]:		outboundproxy
Username [2]:		defaultuser / fromuser@fromdomain
Password:		secret
Authuser [3]:		authuser
	<input checked="" type="checkbox"/> permit outbound calls <input checked="" type="checkbox"/> register	..., register
Language:	de - German (de-DE) ▾	language = de en
Dial() argument [4]:	SIP/{number:1}@{gateway}	~ Dial(SIP/...)
Extract target number from:	invite - Extract target number from `invite` ▾	
Group [5]:	Amt2 ▾	context = from-gg-amt2
Port [6]:	5060 (Default: 5060)	port
Is behind NAT:	<input checked="" type="radio"/> yes <input type="radio"/> no <input type="radio"/> Force rport <input type="radio"/> comedia only (Default: yes)	nat = yes no force_rport comedia
Reroute RTP stream:	no - Don't reroute RTP stream (default) ▾	directmedia = no yes nonat update update,nonat
Monitor availability:	<input checked="" type="radio"/> yes <input type="radio"/> no (Default: yes)	qualify = yes no
Simultaneous calls:	0 (0 for unlimited, Default: 0)	call-limit
DTMF mode:	rfc2833 - RTP meta data ▾ (Default no.: rfc2833)	dtmfmode
Insecure:	invite - no authentication on incoming invites ▾	insecure = no port invite port,invite
Update Remote-Party-ID:	no - Deactivated (default) ▾	sendrpid = no yes pai
Trust Remote-Party-ID:	no - Deactivated (default) ▾	trustrpid = no yes
Codecs:	<input checked="" type="checkbox"/> G.711a <input type="checkbox"/> G.711u <input type="checkbox"/> GSM <input type="checkbox"/> H.261 <input type="checkbox"/> H.263 <input type="checkbox"/> H.263+ (Default: G.711a)	allow
Permit IP subnet [7]:	0.0.0.0/0 (Default: 0.0.0.0/0)	permit

Option	Wert
Registrar/Server	sip.peoplefone.{de, ch, at}
Username	SIP Benutzer
Password	SIP Passwort
Extract target number from	invite
Insecure	invite
Codecs	G.711a