Webitel Asterisk

asterisk (13) webitel. webitel 33 4 . 4- asterisk:

asterisk webitel:

1. SIP Webitel:

sip.conf

```
[webitel]
type=friend
host=dynamic
port=5080
username=webitel
secret=webitel-secret-pss
disallow=all
allow=alaw
allow=ulaw
insecure=invite,port
canreinvite=no
trustrpid=yes
sendrpid=yes
context=from-webitel
```

2. Webitel asterisk:

extensions.conf

```
[from-webitel]
exten => _X.,1,Dial(SIP/${EXTEN})
exten => _X.,n,Hangup
```

3. webitel, public , 100100, SIP , .
   webitel, 33 4 :

extensions.conf

```
exten => _33XX,1,SIPAddHeader(X-Webitel-To:${EXTEN})
exten => _33XX,2,Dial(SIP/webitel/100100)
exten => _33XX,n,Hangup
```

4. asterisk webitel :

5. **default** . 4- asterisk:
Number: **^(\d{4})$**

default callflow

```
[
	{
		"ringback": {
			"call": {
				"name": "$${ru-ring}",
				"type": "tone"
			},
			"transfer": {
				"name": "$${ru-ring}",
				"type": "tone"
			}
		}
	},
	{
		"recordSession": {
			"action": "start",
			"type": "mp3",
			"stereo": false
		}
	},
	{
		"setVar": [
			"hangup_after_bridge=true"
		]
	},
	{
		"bridge": {
			"codecs": [
				"PCMA",
				"PCMU"
			],
			"endpoints": [
				{
					"type": "sipGateway",
					"dialString": "&reg0.$1",
					"name": "asterisk",
					"parameters": [
						"origination_caller_id_number=${caller_id_number}"
					]
				}
			]
		}
	},
	{
		"hangup": ""
	}
]
```

6. **public** . 100100 SIP webitel.
Number: **100100**

public callflow

```
[
	{
		"if": {
			"expression": "${sip_h_X-Webitel-To}",
			"then": [
				{
					"goto": "default:${sip_h_X-Webitel-To}"
				}
			],
			"else": [
				{
					"hangup": "INCOMING_CALL_BARRED"
				}
			]
		}
	}
]
```

7. .

Webitel. on-site SIP .

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- SIP 3CX Phone 6 Webitel on-demand
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