

Teles N720 settings

N710 Settings - Teles platform

The following settings are used when testing on the Teles platform.

The Domain and Outbound server are replaced with dummy values.

Provider settings

In the device go to: **Settings - VoIP providers**

The screenshot shows the 'Profile Download' settings for a VoIP provider. The settings are as follows:

Setting	Value
Provider	Teles
Profile Version	Select VoIP Provider
General data for your service provider	
Domain	Domain
Proxy server address	Domain
Proxy server port	5060
Registration server	Domain
Registration server port	5060
Registration refresh time	180 sec
Network data for your service provider	
STUN enabled	<input checked="" type="radio"/> Yes <input type="radio"/> No
STUN server address	
STUN server port	3478
STUN refresh time	240 sec
NAT refresh time	20 sec
Outbound proxy mode	<input checked="" type="radio"/> Always <input type="radio"/> Automatic <input type="radio"/> Never
Outbound server address	OutboundServer
Outbound proxy port	5082

Device settings

In the device go to: **Settings - Mobile devices**

Advanced VoIP Settings

In the device go to: **Settings - Advanced VoIP Settings**



- N710 Settings - Teles platform
 - Provider settings
 - Device settings
 - Advanced VoIP Settings

Personal Provider Data

A separate SIP connection must be assigned to each handset.

Authentication name:

Authentication password:

Username:

Display name:

Select VoIP provider:

Hide Advanced Settings

Online directories

You can decide which directory will be opened by pressing the directory key and the ENT key on your handset. One online directory can be selected for an automatic name search.

Directory for direct access:

Corporate directory for ENT key:

Automatic look-up:

Network Mailbox Configuration

Call number or SIP name (DN):

Activate network mailbox: Yes No

Apply changes for all SIP connections:

Settings for codecs

Selected codecs	Available codecs
G.722 G.711 a law G.711 u law G.725	G.729

Buttons: < Add, Remove >, Up, Down

SIP over WebRTC connections

Automatic negotiation of GMP Yes No
Transmission

G.722 codec

Enabling or disabling the G.722 codec will restart the system.
Connections with Icecast2 will be terminated.

Enable wideband via codec Yes No
G.722

One-time station enables a maximum of 4 wideband calls.

G.729 codec

A connection to the internet is necessary for activating a G.729 license.

Activate G.729 codec

Call Transfer

Use the R key to initiate call transfer with the SIP Refer method Yes No

Transfer Call by On Hook Yes No

You can define the choice of target address in the SIP protocol

Prefer target address automatically Yes No

Hold on transfer target For attended transfer
 For unattended transfer

Hook Flash (R key)

R key settings are disabled because the R key is being used for call transfer.

Listen ports for WebRTC connections

Use random ports for SIP Yes No

SIP port -