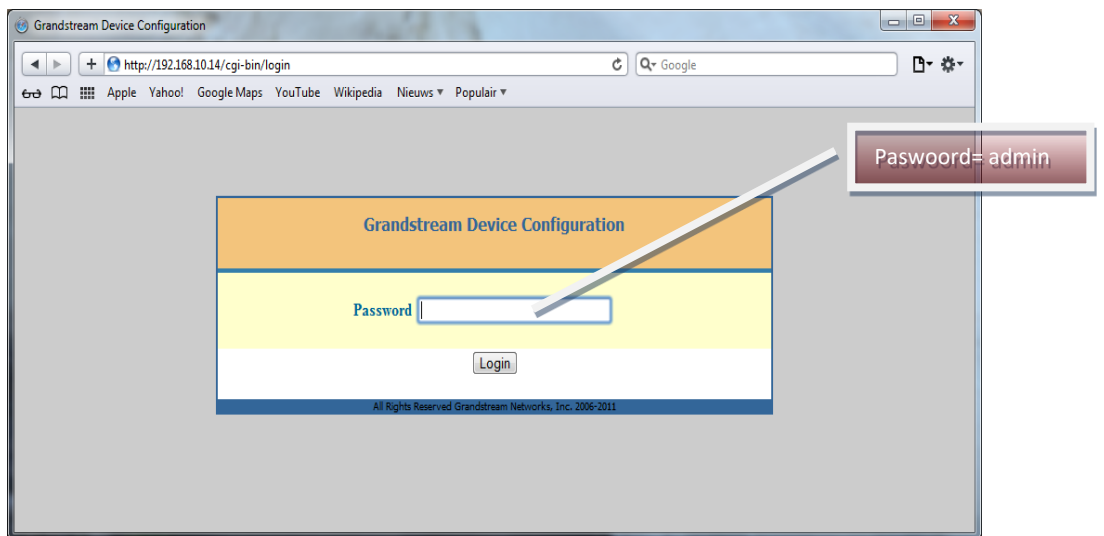


IPTALK ed2.0 icm HT701

- Na aansluiten van de IPTALK dient men eerst te bepalen of hij via DHCP of met een statisch IP adres moet werken. Standaard is dit DHCP.
 - Hoe wijzigen?
 - Sluit analoog toestel aan, neem hoorn op en vorm **** kies daarna de optie 01. De IPTalk vermeldt met een tekstbericht welke instelling actief is. Met 9 kan men wisselen tussen statisch of dynamisch IP-adres.
Met * ga je naar het volgende menu
 - Via optie 02 kan men eventueel het statische IPadres ingeven (9 cijfers bv: 192168013004 staat voor 192.168.13.4)
 - Via optie 03 kan men subnetmask ingeven (bijv: 255255255000)
 - Via optie 04 kan men de default gateway ingeven (enkel benodigd voor eventuele latere update van de software)
- Nadat het IP adres ingegeven is kan men de IPTALK via de webbrowser benaderen
 - `http://<ipadres IPTALK>`



- Onderstaande instellingen **in rood omkaderd** moeten gemaakt worden in **ADVANCED SETTINGS**

Grandstream Device Configuration

STATUS
BASIC SETTINGS
ADVANCED SETTINGS
FXS PORT

Admin Password: (purposely not displayed for security protection)

Layer 3 QoS: 48 (Diff-Serv or Precedence value)

Layer 2 QoS: 802.1Q/VLAN Tag 0 802.1p priority value 0 (0-7)
802.1p priority value (NATed traffic) 0 (0-7)

STUN server is : (URI or IP:port)

Keep-alive Interval: 20 (in seconds, default 20 seconds)

Use STUN to detect network connectivity: No
 Yes, total STUN response misses 3 to restart DHCP (minimum=3)

Firmware Upgrade and Provisioning: Upgrade Via TFTP HTTP HTTPS
Firmware Server Path:
 Config Server Path:

XML Config File Password:
 HTTP/HTTPS User Name:
 HTTP/HTTPS Password:

Firmware File Prefix: Firmware File Postfix:
 Config File Prefix: Config File Postfix:

Allow DHCP Option 66 to override server:
 No Yes

Automatic Upgrade:
 No Yes, every 10080 minutes(60-5256000).
 Yes, daily at hour 1 (0-23). Yes, weekly on day 1 (0-6).

Always Check for New Firmware at Boot up
 Check New Firmware only when F/W pre/suffix changes
 Always Skip the Firmware Check

Authenticate Conf File: No Yes (cfg file would be authenticated before acceptance if set to Yes)

Firmware Key: (in Hexadecimal Representation)

SSL Certificate:

SSL Private Key:

SSL Private Key Password:

ACS URL:

ACS Username:
ACS Password:
Periodic Inform Enable: No Yes
Periodic Inform Interval:
Connection Request Username:
Connection Request Password:
System Ring Cadence:
Dial Tone:
Ringback Tone:
Busy Tone:
Call Progress Tones: Reorder Tone:
Confirmation Tone:
Call Waiting Tone:
Syntax: f1=val [, f2=val [, c=on1/off1 [-on2/off2 [-on3/off3]]]]; (Frequencies are in Hz and cadence on and off are in ms)
Lock Keypad Update: No Yes (configuration update via keypad is disabled if set to Yes)
Disable Voice Prompt: No Yes (voice prompt is disabled if set to Yes)
Disable Direct IP Call: No Yes (direct IP call is disabled if set to Yes)
NTP Server: (URI or IP address)
Allow DHCP option 42 to override NTP server: No Yes
Syslog Server:
Syslog Level: NONE
Send SIP Log: No Yes
Download Device Configuration:
Upload firmware:

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Klik vervolgens op Update en men bekomt volgend scherm:

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT

Your configuration changes have been saved.
They will take effect on next reboot.

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Kies vervolgens **FXS PORT**

SIP server en Outbound Proxy: Ipadres van de SIP server.
SIP UserID en Name: intern telefoonnummer van de IPTALK.

Grandstream Device Configuration

STATUS	BASIC SETTINGS	ADVANCED SETTINGS	FXS PORT
Account Active: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Primary SIP Server: <input type="text" value="192.168.10.251"/> (e.g., sip.mycompany.com, or IP address)			
Failover SIP Server: <input type="text"/> (Optional, used when primary server no response)			
Prefer Primary SIP Server: <input checked="" type="radio"/> No <input type="radio"/> Yes (yes - will register to Primary Server if Failover registration expires)			
Outbound Proxy: <input type="text" value="192.168.10.251"/> (e.g., proxy.myprovider.com, or IP address, if any)			
SIP Transport: <input checked="" type="radio"/> UDP <input type="radio"/> TCP <input type="radio"/> TLS (default is UDP)			
NAT Traversal (STUN): <input checked="" type="radio"/> No <input type="radio"/> No, but send keep-alive <input type="radio"/> Yes			
SIP User ID: <input type="text" value="81"/> (the user part of an SIP address)			
Authenticate ID: <input type="text"/> (can be identical to or different from SIP User ID)			
Authenticate Password: <input type="text"/> (purposely not displayed for security protection)			
Name: <input type="text"/> (optional, e.g., John Doe)			
DNS Mode: <input checked="" type="radio"/> A Record <input type="radio"/> SRV <input type="radio"/> NAPTR/SRV			
Tel URI: <input type="text" value="Disabled"/>			
SIP Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Unregister On Reboot: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Outgoing Call without Registration: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Register Expiration: <input type="text" value="60"/> (in minutes. default 1 hour, max 45 days)			
SIP Registration Failure Retry Wait Time: <input type="text" value="20"/> (in seconds. Between 1-3600, default is 20)			
Local SIP port: <input type="text" value="5060"/> (default is 5060 for UDP and TCP; 5061 for TLS)			
Local RTP port: <input type="text" value="5004"/> (1024-65535, default 5004)			
Use Random Port: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Refer-To Use Target Contact: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Transfer on Conference Hangup: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Enable Ring-Transfer: <input checked="" type="radio"/> No (RFC5589 Semi-Attended Transfer) <input type="radio"/> Yes			
Disable Bellcore Style 3-Way Conference: <input checked="" type="radio"/> No <input type="radio"/> Yes (Using star code *23 for 3-way conference)			
Remove OBP from Route Header: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Support SIP Instance ID: <input type="radio"/> No <input checked="" type="radio"/> Yes			
Validate Incoming SIP Message: <input checked="" type="radio"/> No <input type="radio"/> Yes			
Check SIP User ID for incoming INVITE: <input checked="" type="radio"/> No <input type="radio"/> Yes (no direct IP calling if Yes)			
Allow Incoming SIP Messages from SIP Proxy Only: <input checked="" type="radio"/> No <input type="radio"/> Yes (no direct IP calling if Yes)			
SIP T1 Timeout: <input type="text" value="0.5 sec"/>			
SIP T2 Interval: <input type="text" value="4 sec"/>			
DTMF Payload Type: <input type="text" value="101"/>			
Preferred DTMF method: Priority 1: <input type="text" value="In-audio"/>			
(in listed order) Priority 2: <input type="text" value="In-audio"/>			
Priority 3: <input type="text" value="In-audio"/>			
Disable DTMF Negotiation: <input checked="" type="radio"/> No (negotiate with peer) <input type="radio"/> Yes (use above DTMF order without negotiation)			
Send Hook Flash Event: <input checked="" type="radio"/> No <input type="radio"/> Yes (Hook Flash will be sent as a DTMF event if set to Yes)			

Enable Call Features: No Yes (if Yes, call features using star codes will be supported locally)

Offhook Auto-Dial: (User ID/extension to dial automatically when offhook)

Proxy-Require:

Use NAT IP: (used in SIP/SDP message if specified)

Distinctive Ring Tone: Ring Tone 1 used if incoming caller ID is
Ring Tone 1 used if incoming caller ID is
Ring Tone 1 used if incoming caller ID is

Disable Call-Waiting: No Yes

Disable Call-Waiting Caller ID: No Yes

Disable Call-Waiting Tone: No Yes

Disable Receiver Offhook Tone: No Yes (ROH tone will not be played after offhook for 60 seconds)

Disable Reminder Ring for On-Hold Call: No Yes

Disable Visual MWI: No Yes

Ring Timeout: 60 (10-300, default is 60 seconds)

Delayed Call Forward Wait Time: 20 (Allowed range 1-120, in seconds.)

No Key Entry Timeout: 1 (in seconds, default is 4 seconds)

Early Dial: No Yes (use "Yes" only if proxy supports 484 response)

Dial Plan Prefix: (this prefix string is added to each dialed number)

Use # as Dial Key: No Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)

Dial Plan: {x+!*x+}

SUBSCRIBE for MWI: No, do not send SUBSCRIBE for Message Waiting Indication
 Yes, send periodical SUBSCRIBE for Message Waiting Indication

Send Anonymous: No Yes (caller ID will be blocked if set to Yes)

Anonymous Call Rejection: No Yes

Special Feature: Standard

Session Expiration: 180 (in seconds, default 180 seconds)

Min-SE: 90 (in seconds, default and minimum 90 seconds)

Caller Request Timer: No Yes (Request for timer when making outbound calls)

Callee Request Timer: No Yes (When caller supports timer but did not request one)

Force Timer: No Yes (Use timer even when remote party does not support)

UAC Specify Refresher: UAC UAS Omit (Recommended)

UAS Specify Refresher: UAC UAS (When UAC did not specify refresher tag)

Force INVITE: No Yes (Always refresh with INVITE instead of UPDATE)

Send Re-INVITE After Fax: No Yes

Enable 100rel: No Yes

Use First Matching Vocoder in 200OK SDP: No Yes

Preferred Vocoder: choice 1: PCMA
(in listed order) choice 2: PCMA
choice 3: G723
choice 4: G729

G723 Rate: 6.3kbps encoding rate 5.3kbps encoding rate
 iLBC Frame Size: 20ms 30ms
 iLBC Payload Type: (between 96 and 127, default is 97)
 VAD: No Yes
 Symmetric RTP: No Yes
 Fax Mode: T.38 Pass-Through
 Re-INVITE After Fax Tone Detected: Enabled Disabled
 Jitter Buffer Type: Fixed Adaptive
 Jitter Buffer Length: Low Medium High
 SRTP Mode: Disabled Enabled but not forced Enabled and forced

SLIC Setting: EUROPEAN CTR21
Caller ID Scheme: ETSI-FSK during ringing
Polarity Reversal: No Yes (reverse polarity upon call establishment and termination)

Loop Current Disconnect: No Yes (loop current disconnect upon call termination)
 Loop Current Disconnect Duration: (100 - 10000 milliseconds. Default 200 milliseconds)
 Hook Flash Timing: In 40-2000 milliseconds range, minimum: maximum:
 On Hook Timing: (In 40-2000 milliseconds range, default is 400)
 Gain: TX RX
 Disable Line Echo Cancellor (LEC): No Yes

Ring Tones (Syntax: c=on1/off1-on2/off2-on3/off3;)
 Ring Tone 1:
 Ring Tone 2:
 Ring Tone 3:
 Ring Tone 4:
 Ring Tone 5:
 Ring Tone 6:
 Ring Tone 7:
 Ring Tone 8:
 Ring Tone 9:
 Ring Tone 10:

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Klik vervolgens op **Update** en men bekomt volgend scherm.

Grandstream Device Configuration

Your configuration changes have been saved.
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Klik vervolgens op **BASIC SETTINGS** en men bekomt volgend scherm.

Grandstream Device Configuration

End User Password: (purposely not displayed for security protection)

Web Port: (default for HTTP is 80)

Telnet Server: No Yes

IP Address: dynamically assigned via DHCP

DHCP hostname: (optional)

DHCP domain: (optional)

DHCP vendor class ID: (optional)

use PPPoE

PPPoE account ID:

PPPoE password:

PPPoE Service Name:

Preferred DNS server:

statically configured as:

IP Address:

Subnet Mask:

Default Router:

DNS Server 1:

DNS Server 2:

Time Zone:

Self-Defined Time Zone: (For example: MTZ+6MDT+5,M4.1.0,M11.1.0)

Allow DHCP server to set Time Zone: No Yes

Language:

Reset Type:

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Hier kan eventueel nog de instellingen gemaakt worden indien een statisch IPadres gewent is.

Kies vervolgens op **Reboot**.

De IPTALK is gereed voor gebruik met uw Fasttel deurtelefoon. Deze laatste dient wel nog geprogrammeerd te worden met oa een oproepnummer