Technical Training
Deerfield.com

Agenda

- **High level overview of SIP & RTP**
- **Call set-up, ports needed and codecs**
- **SIP phone configuration**
- **VOIP Gateway configuration**
- **Server status Log**
- **Partner support procedures**

What is 3CX Phone System?

- **IP PBX based on the SIP standard**
- **Works with popular SIP phones, VOIP providers and VOIP Gateways**
- **Includes:**
  - a SIP server for call set-up
  - a media server for managing audio streams
  - an IVR server for Auto Attendant and Voice Mail
  - Web server (apache)
  - Database server (Postgres)
- **Runs on Windows XP / Vista / 2000 / 2003**

Benefits of 3CX Phone System

- **No need for separate phone wiring**
- **Scalable**
  - Add PSTN lines by adding a VOIP Gateway or VOIP Provider
  - Add extensions by plugging in additional phones
- **Extension tied to phone, not network point – enables roaming and teleworking**

SIP Standard – What is it?

- **Session Initiation Protocol**
- **Text based - similar to the SMTP/HTTP protocol**
- **Sets up phone calls only**
- **Supported by all major equipment & software manufacturers -** mix and match IP software & hardware from different vendors (Cisco, Linksys, Patton Grandstream, etc)
- **Defined in RFC 3261** http://tools.ietf.org/html/rfc3261

What is RTP?

- Real Time Transport Protocol
- Defines format for delivering audio and video over the internet
- Each call requires 2 RTP channels, one for each phone, aka “Endpoint”
- Defined in RFC 1889 http://tools.ietf.org/html/rfc1889

Common SIP Requests

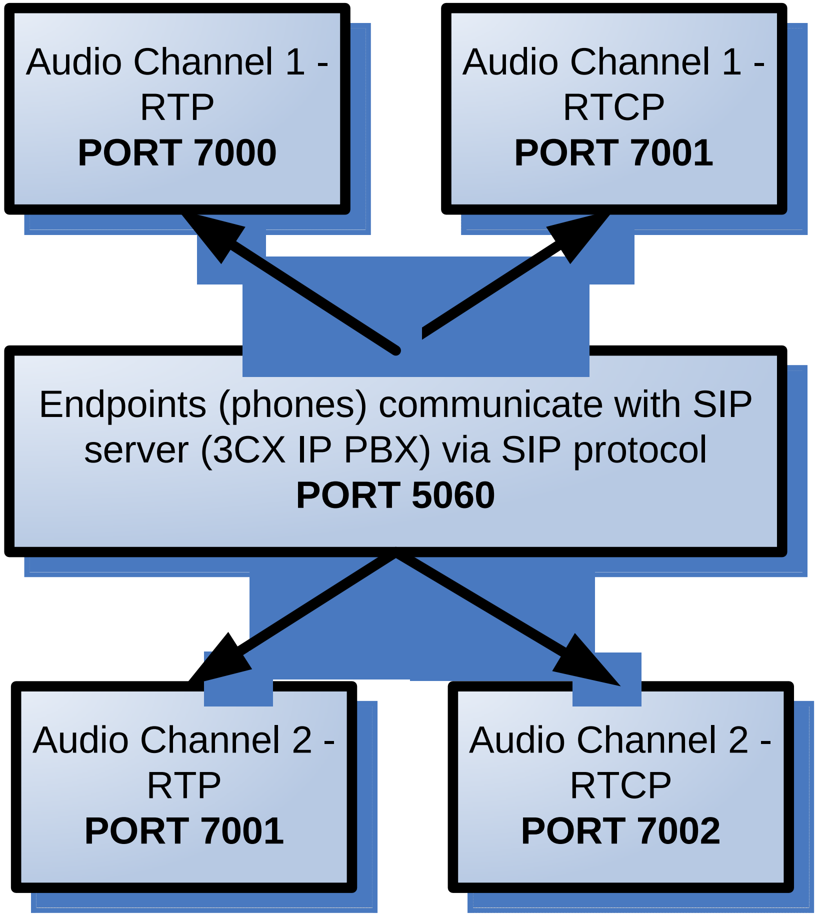
- **REGISTER – to register a phone or line with a SIP server**
- **INVITE – to set-up a call**
- **CANCEL – to cancel a call set-up**
- **BYE – to terminate a call**

Common SIP responses

- **100 trying**
- **180 ringing**
- **200 OK**
- **401 not authorized**
- **404 destination not found**
- **486 busy**
- **Note similarity to HTTP responses!**
- **More detail about SIP in technical manual at:** http://www.3cx.com/downloads/techmanual31.pdf

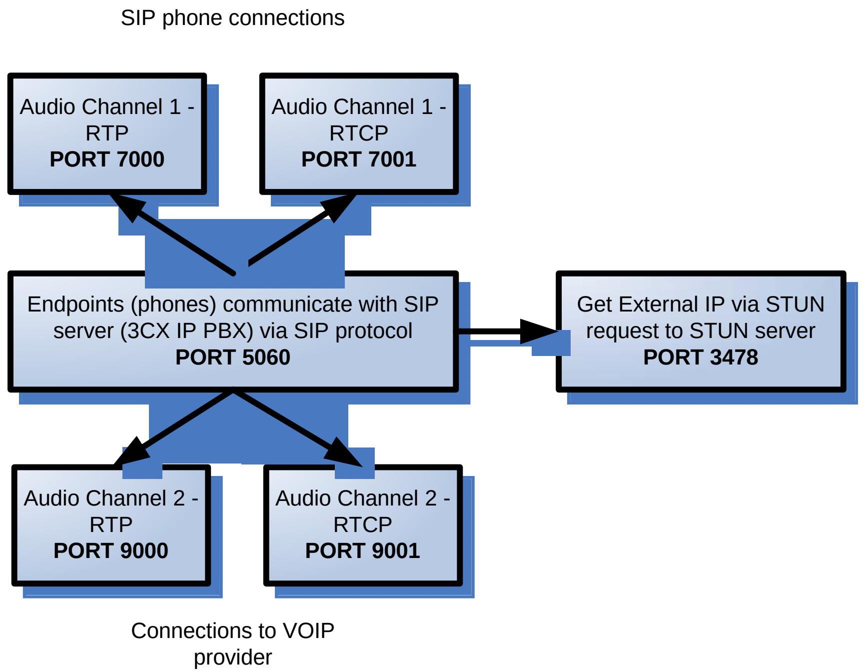
Typical Call setup – Internal call

- **Call setup for a call to another extension or via a VOIP gateway**
- **Port range for RTP is 7000- and up (configurable)**
- **2 Ports per Endpoint, one for RTP and one for RTCP!**



Typical Call setup – External call

- **Call setup for a call via a VOIP provider or an external extension**
- **Port range for RTP to VOIP provider is 9000- and up (configurable)**
- **2 Ports per endpoint, one for RTP and one for RTCP!**
- **STUN request needed to get external IP**



Codecs

- **Codecs encode/compress the audio stream**
- **3CX supports**
  - G711 ulaw, mlaw (approx 80 kb per second)
  - GSM (approx 20-30 kb per second)
  - Speex (approx 20-30 kb per second)
- **We do not support**
  - G729 (does not support in-band DTMF)
  - G723
  - Both have high licensing costs, and several other large market players (such as Microsoft) are not including these codecs in their products

Configuring SIP phones & VOIP Gateways

SIP phones

- 3CX supports both hardware and software SIP phones from all popular vendors
- In theory, we should work with all SIP phones, in practice its better to use tested phones.
- Supported phones are listed here:

http://www.3cx.com/support/sip-phones.html

SIP phone configuration

- **Create extension in 3CX Management Console**
- **Take note of these details**
  - Extension number
  - Authentication ID
  - Authentication Password
  - FQDN / IP of 3CX Phone System
- **Configure these details on the phone**

Example: SNOM 360

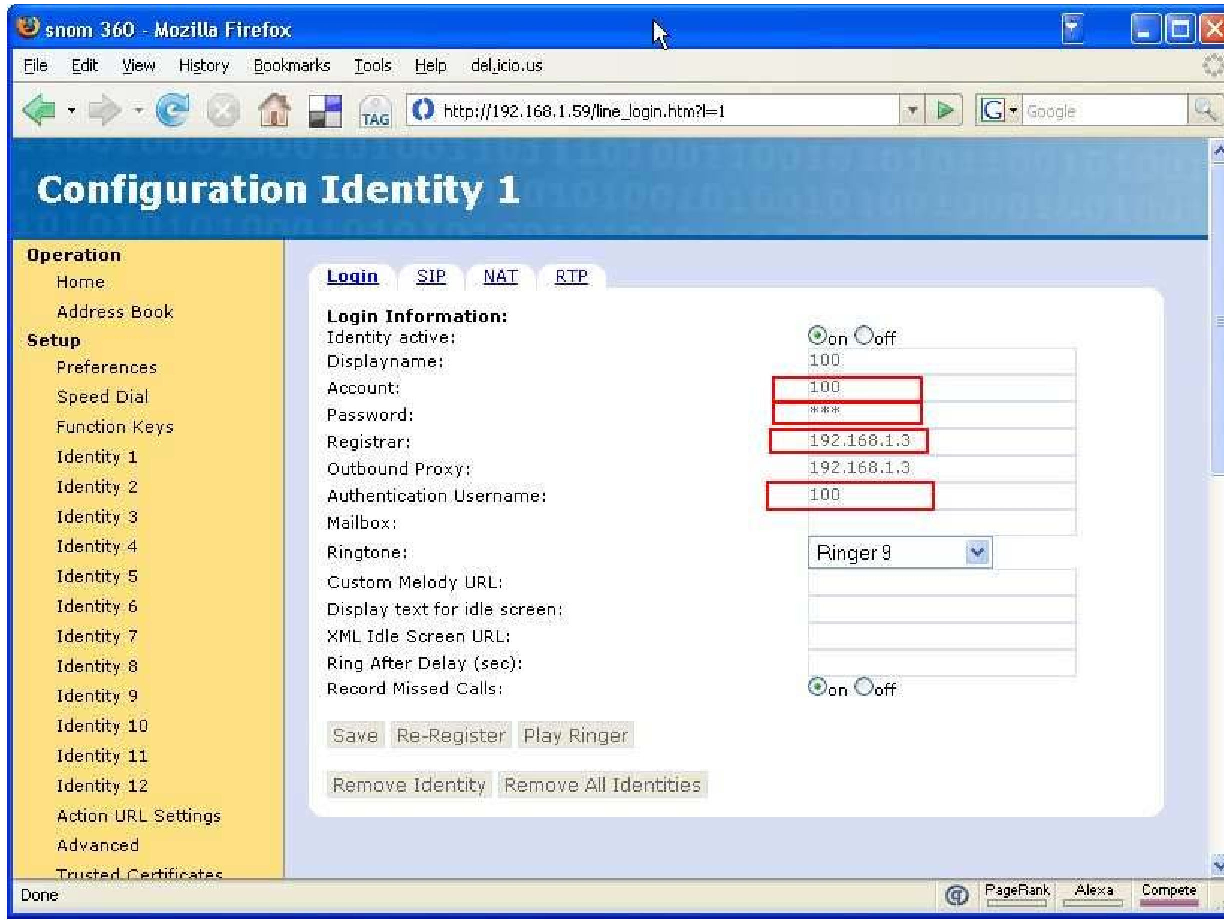
Extension No = Account

Password = Authentication Password

Registrar = IP of 3CX Phone System

Authentication Username = Authentication ID

Each vendor has its own terminology!



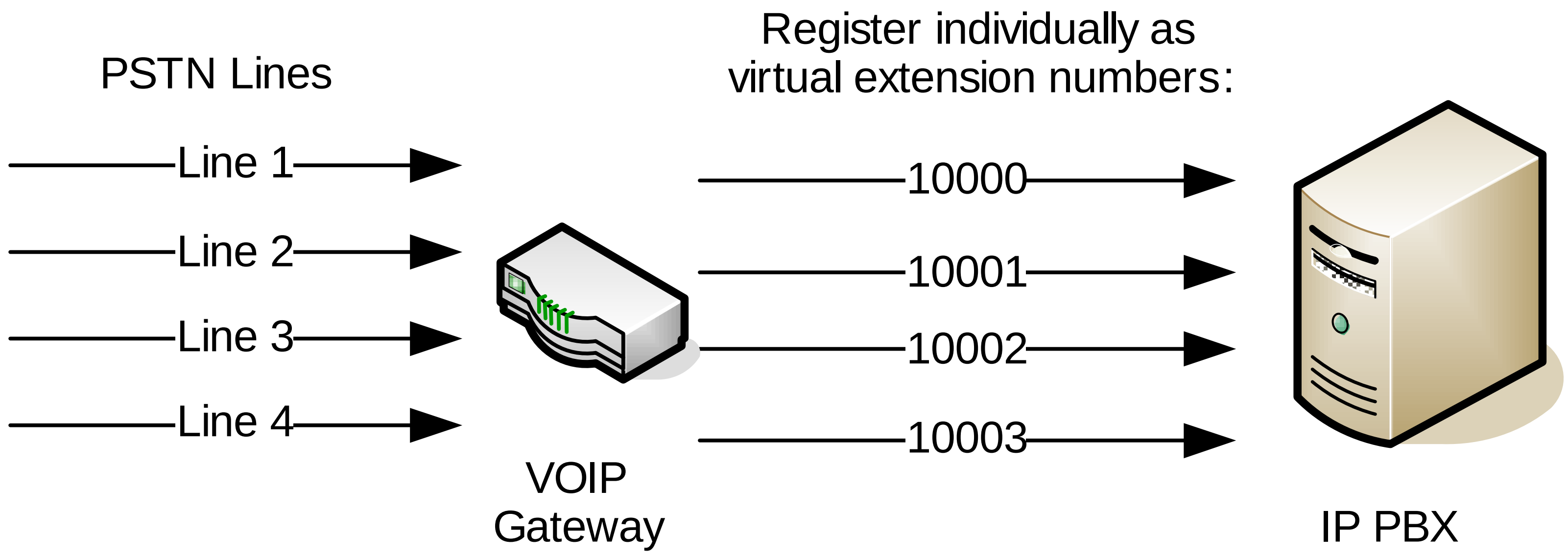
VOIP Gateways

- VOIP gateways aka FXO gateways convert PSTN lines to SIP extensions
- Each line on a VOIP Gateway is a SIP extension – configuration is similar
- Gateway is separate network device with own IP – It can be anywhere on your network
- Supported gateways are listed here:

http://www.3cx.com/support/voip-gateways.html

VOIP Gateway configuration overview

- **Each line on a VOIP Gateway is associated with a virtual extension on 3CX Phone System:**

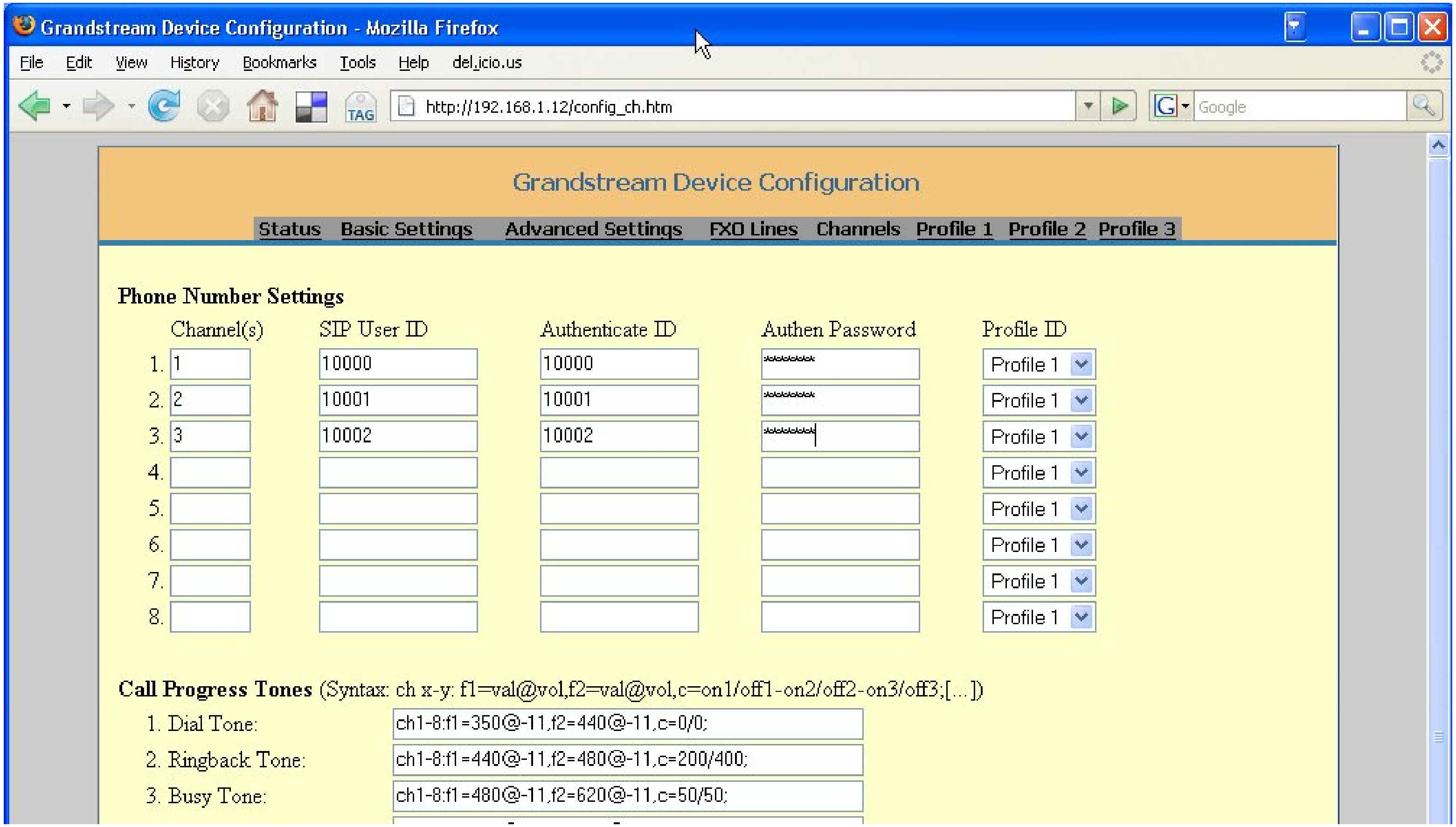


VOIP Gateway configuration

- **To create PSTN lines, click ‘Add PSTN’ in 3CX Management console.**
- **When you click finish, the following details will be provided:**
  - IP of 3CX Phone System
  - Virtual extension number for each line
  - Authentication ID for each line
  - Authentication Password for each line
- **Enter these into the Gateway**
- **Configuration guides for popular gateways: http://www.3cx.com/voip-gateways/index.html**

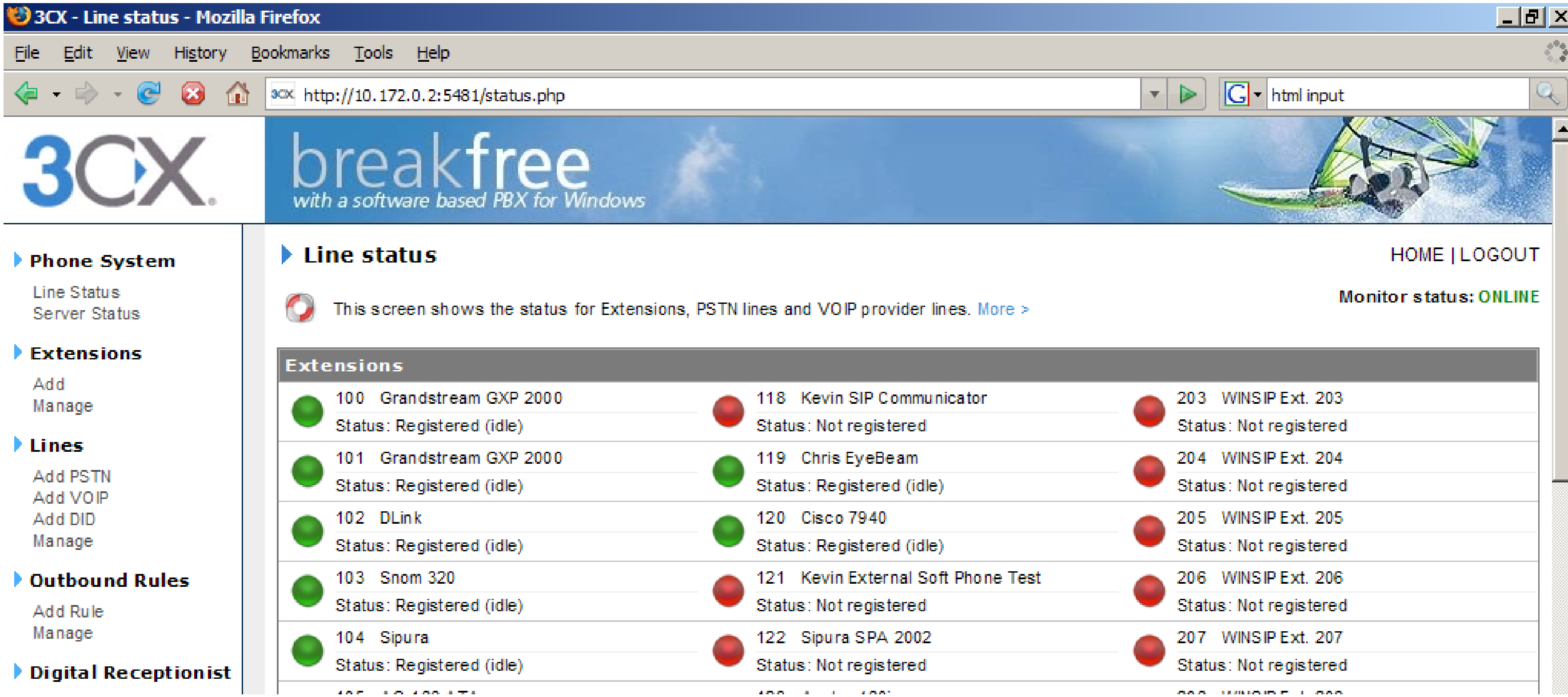
Example: Grandstream GXW 4104/8

- **Sip user ID, Auth ID, Auth Pw inserted in channels page**



Check registration in Line Status screen

- **Switch to Line Status in 3CX Management console to see if all lines registered OK**



VOIP providers

- **3CX works with mainstream SIP VOIP providers:**

  http://www.3cx.com/support/voip-providers.html

- **Only G711, GSM, Speex codecs are supported**
- **We will test against new main stream VOIP providers – give us authentication details and 1-2 weeks and we will provide results, with reason of failure if not compatible / recommended.**

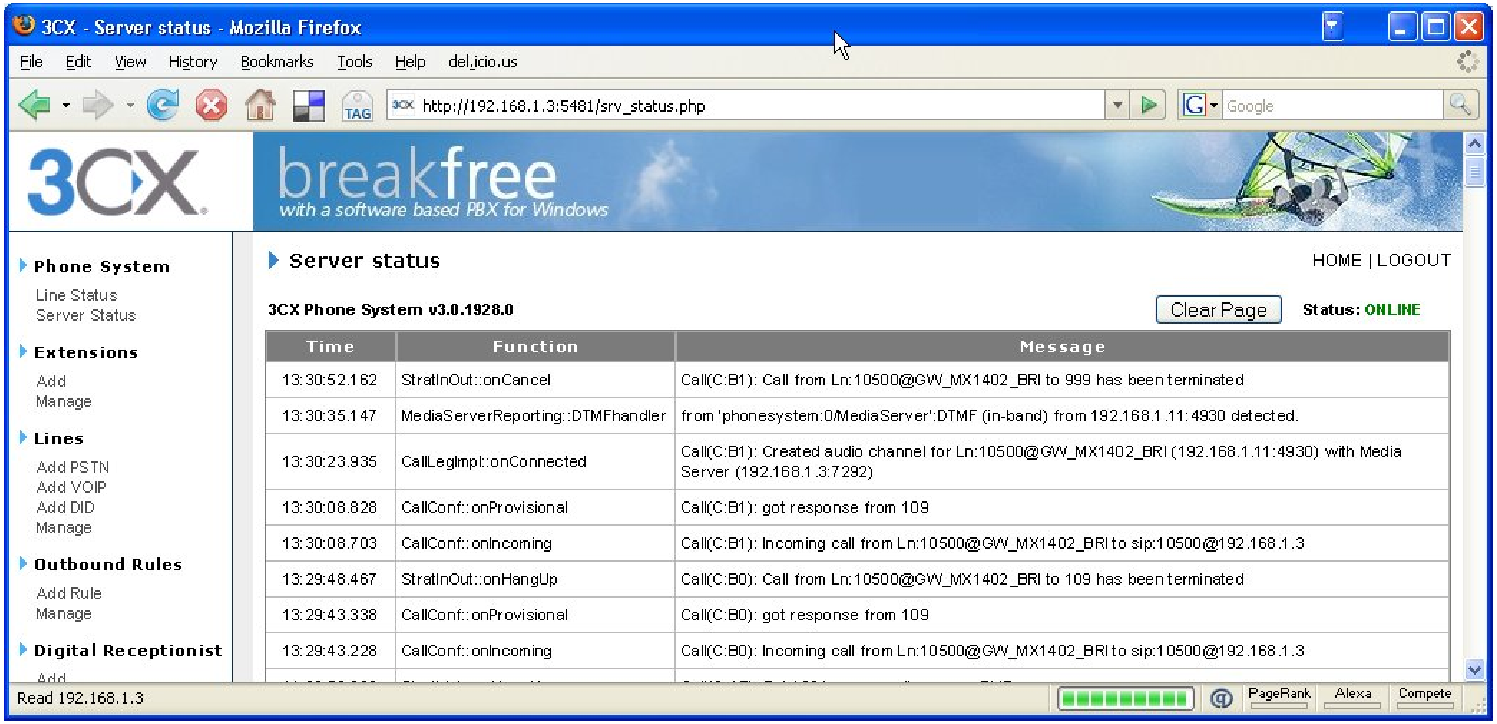
3CX Troubleshooting

Most common problems

- **Internet-Facing Firewall configuration**
  - Port 5060 open for SIP
  - Port 3478 open for STUN
  - Port 9000- open for RTP (if using VOIP Provider)
- **Registration failures**
  - Use of wrong authentication ID – Password
  - Extension number not inserted in right field
- **Use of unsupported phones, providers or gateways**
- **Use server status log to analyze problem**

Server status log

- **3CX Management Console : Phone System > Server status**



Server status log (2)

- **Logs important messages and errors**
  - Call(C:B1): Call from Ln:10500@GW_MX1402_BRI to 999 has been terminated
  - 13:30:35.147 DTMF (in-band) from 192.168.1.11:4930 detected.
  - 13:30:23.935 Call(C:B1): Created audio channel for Ln:10500@GW_MX1402_BRI (192.168.1.11:4930) with Media Server (192.168.1.3:7292)
  - 13:30:08.828 Call(C:B1): got response from 109
  - 13:30:08.703 Call(C:B1): Incoming call from Ln:10500@GW_MX1402_BRI to sip:10500@192.168.1.3
- **In this example:**
  - 192.168.3 is SIP server, GW_MX1402_BRI is gateway with IP 192.068.1.11.
  - Call with ID B1 has been received on virtual line 10500, routed to extension 109, then after the default 15-second timeout, sent to voice mail by the gateway from port 4930 to the media server on port 7292 (meaning the extension is internal), because no answer was received from extension 109.

Server status log (3)

- Server status log shows IP of SIP server, SIP phones, gateways and VOIP providers
- Shows on which ports audio channels are established (e.g. 192.168.1.3:7292)
- Shows whether extensions are on same subnet
- Gives reasons if phone, provider or gateway fail to register
- Allows you to more easily troubleshoot network and firewall configuration issues
- Always include server status log with support request!

Technical manual

- **More detail about server status log messages in the technical manual at**
  http://www.3cx.com/downloads/techmanualv31.pdf

Debug logs

- **Problems during setup, review**
  - C:\Program Files\3CX PhoneSystem\install.log
- **Problems with database access**
  - C:\Program Files\3CX PhoneSystem\Data\DB\pg_log*.log
- **Problems with apache web server**
  - C:\Program Files\3CX PhoneSystem\Bin\Apache\logs*.log
- **Problems with SIP server**
  - C:\Program Files\3CX PhoneSystem\Data\Logs\3CXPhoneSystem*.log
- **Problems with Media server**
  - C:\Program Files\3CX PhoneSystem\Data\Logs\3CXMediaServer*.log

Debug logs (2)

- **VoiceMail:**
  - C:\Program Files\3CXPhoneSystem\Data\Logs\3CXVoiceBoxManager*.log
- **IVR:**
  - C:\Program Files\3CX PhoneSystem\Data\Logs\3CXIvrServer*.log
  - C:\Program Files\3CX PhoneSystem\Data\Logs\IvrPhp.log
- **After each phone system service restart, logs are backed up to**
  - C:\Program Files\3CX PhoneSystem\Data\Logs\Backup

3CX Phone System – Versions

- **Free edition**
  - Up to 8 configured lines, no support
- **Small Business Edition**
  - Up to 16 configured lines and 25 extensions
  - Full version of Call Assistant (call transfer, call park & pickup, etc.) & Message Waiting Indicator (MWI)
  - Ability to buy support and maintenance
  - Price $350

3CX Phone System – Versions

- **Pro Edition**
  - Up to 32 configured lines, UNL extensions
  - Advanced call assistant & MWI
  - Price $895
- **Enterprise Edition**
  - Unlimited extensions and lines
  - Exchange Server 2007 integration
  - Call Queuing
  - Cluster ready
  - Price $1250

3CX Support Procedures

Support procedures

- **Resellers receive priority technical support to assist in servicing your customers quickly and efficiently**
- **Login to your account at https://shop.deerfield.com/resellers/**
- **Submit support incident with**
  - SIP phones, VOIP Provider and VOIP Gateways used
  - Detailed problem description, eg: transfer fails when receiving call from VOIP Provider XYZ
  - Include Server Status log
- **Reseller support available 9:00AM to 5:00PM EST**

Thank you

- **Thank you for attending this technical presentation**

- **For questions, please don't hesitate to email us resellersupport@deerfield.com**