



# Virtual WYDE Voice Demo Server – Setup Guide

(version 2.1)

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For the most current versions of documentation, go to the WYDE support Web site:  
<http://www.wydevoice.com/support>

April 9, 2010

## Symbols and Notations in this Manual

The following notations and symbols can be found in this manual.



Denotes any item that requires special attention or care. Damage to the equipment or the operator may result from failure to take note of the noted instructions

<b>Figure</b>	Denotes any illustration
<b>Table</b>	Denotes any table
Text	Denotes any text output
<i>Folder/File</i>	Denotes any folders (paths) or files names
commands	Denotes any commands, attributes and parameters

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## Chapter 1: Introduction

The WYDE Voice conferencing bridge software can be provided for demonstration and trial purposes as fully installed and configured demonstration VMware image. This image can be used to create virtual machine within VMware products (such as VMware Player, VMware Server, VMware Workstation, etc.); this guide explains how to use this image within VMware Player. This virtual machine image can be used to trial the WYDE Voice conferencing bridge software.

### *Assumed Skills*

This Virtual WYDE Voice Demo Server Setup Guide assumes you have a working knowledge of the following technologies and skills:

- PC usage
- System administration
- VOIP basics
- TCP/IP networking
- Linux/CentOS basics
- VMware Player basics
- CounterPath X-Lite basics or SJ Labs SJphone basics
- Web Administration Interface – User Guide
- WYDE Software Installation Guide

### *Used Software*

This section explains what additional software you will need to use provided WYDE Voice conferencing bridge software VMware demonstration image and to make test calls to the system from your computer.

### **VMware Player**

To use provided VMware image you will need VMware Player or similar VMware virtualization products (VMware Server, VMware Workstation, etc.); this guide explains how to use this image within VMware Player.

VMware Player is the easiest way to run multiple operating systems at the same time on your PC. It has user-friendly interface and it makes effortless for anyone to try out isolated virtual machines to safely test new software. VMware Player can also be used to run a pre-created virtual copy of a computer with installed software that is provided to you for testing.

VMware Player can be downloaded from VMware web site using URL:  
<https://www.vmware.com/products/player/>

### **CounterPath X-Lite**

To make testing voice calls from your computer to the configured WYDE Voice conferencing bridge you will need any of VoIP softphones. CounterPath X-Lite is one of

such soft phone that uses the Session Initiation Protocol. This guide explains how to use X-Lite software to make test calls to the WYDE bridge.

CounterPath X-Lite is the easiest way to make voice calls, video calls and instant messaging; it helps you seamlessly transition from a traditional phone environment into the world of Voice over IP. There are currently two major releases of X-Lite with radically different interfaces. X-Lite 2.0 for Linux, which uses the old X-Pro code base and X-Lite 3.0 for Windows and Mac OS X which uses the eyeBeam code base. Any version of X-Lite can be used to make VoIP calls the WYDE bridge.

CounterPath X-Lite can be downloaded from CounterPath web site using URL:

<http://www.counterpath.net/x-lite.html>

### **SJ Labs SJphone**

As alternative of X-Lite you can use SJphone produced by SJ Labs to make test calls from your computer to the configured WYDE Voice conferencing bridge.

SJphone is a VOIP softphone that allows you to speak with any other softphone running on a PC/PDA, any stand-alone IP-phone, or using Internet Telephony Service Provider (ITSP) with any traditional wired or mobile phone. It supports SIP and is fully inter-operable with most major VOIP vendors and ITSPs.

SJ Labs SJphone can be downloaded from SJ Labs web site using URL:

<http://www.sjphone.org/sjp.html>

### **Installed Software**

We assume that you have VMware Player and either CounterPath X-Lite or SJ Labs SJphone installed on your computer. If no, please download them from URLs provided above and install the products on your computer.

### ***Definitions, Acronyms and Abbreviations***

In order to discuss the VMware virtual WYDE Voice demo server setup process effectively, we need to have a common set of terminology. For this purpose, we should definite the dictionary for the terms you will see throughout this guide:

- **VoIP** – Voice over Internet Protocol, a term that refers to the capture/playback of audio streams and their transmission over IP based networks.
- **End Point (EP)** – A generic term used to denote the application running on end-user machines in a VoIP.
- **Public Switched Telephone Network (PSTN)** – the traditional phone system.
- **Bridge** – A server that hosts voice conferences. Participants can use PSTN or VoIP connections to connect to the bridge. It is responsible for mixing the signals and sending the result back to the participants.
- **Gateway** – A gateway server between PSTN and VoIP, i.e. a server that terminates end point connections and routes VoIP data between an end point and the bridge.
- **Conference User** – A user in a conference. Each connection to the conference bridge is associated with exactly one conference user. An end point can be associated with any

number of conference users. A conference user may or may not be associated with an end point. The conference user can have one of the roles: host, participant or listener.

- **Conference** – An audio meeting hosted on a bridge and consisting of PSTN and/or VoIP participants. A data structure is used to describe ongoing conference on the bridge. Objects of this type are only created by server. User may fetch these objects by calling appropriate function. When conference is over the conference object is deleted by the server.
- **Conference Number** – A unique external conference number. Conference number is the property of conference account. If the conference accounts have the same conference number all these accounts determine one single conference. For instance the user can create one conference account record that determine host role, another conference account record that determine participant role, and another conference account record that determine listener role – all these records should have the same conference number to determine one unique conference.
- **Conference ID** – A unique conference ID that represents the instance of a conference. When any conference is being started it receives unique conference ID, and all calls to this conference have the same conference ID; if this conference has been completed and another conference is being started that conference will receive another conference ID. Conference ID is normally not exposed to users, unless on the reports.
- **Session** – A data structure represents a single ongoing call on the server. User can not directly create this object. When the call is over server automatically deletes this object. Normally this data structure is used to get information about call attributes like calling/called number etc., or do something with the call, for instance mute, hang, hold etc.
- **Session ID** – The unique identifier generated by the bridge for each session (connection, VoIP as well as PSTN) established between a conference user and the bridge. The session id is unique within a given conference.
- **Audio Key** – A key sequence that is used to group different calls from the same conference in a bundle to manage these calls using real-time or another external interface. Audio key is short identifier generated externally and provided to the bridge at the time of joining a conference. Audio key is being generated by real-time application, for instance Moderator-Console, the user can enter the same audio key on his DTMF keypad, usually as #audio key#, these calls (the call from real-time application and the user call to the conference) are being grouped together and the real-time application can manage this user call (the call with the same audio key), for instance mute the call, etc.
- **Subscriber** – A real person, he has a name, phone number, e-mail address, etc. The subscriber can have conference accounts, he does not have access codes, but access codes are properties of conference accounts that have subscribers. Note that non-admin (non-operator) subscribers can see only “own” information, i.e. his information and information that belongs to subscribers created by him, he can see only their calls, conferences, the reports will show only their data, etc.
- **PIN** – The login ID for the subscriber (must be unique). It can be used either as login in Web Administration Interface (in this case it can be either number or alpha-numeric) or as login for some call flows (in this case must be numeric) for participants authorization.



- **Conference Account** – The element of subscriber conferences configuration. Conference accounts always belong to subscriber. It is being used to define a person in a conference with a particular role (e.g. host, participant, listener, etc.), the DNIS number that should be used to call to the conference, and the access code that should be entered by the user that called to the conference DNIS to determine his role. A subscriber could be a host user in one conference and a listener in another. Conference accounts with the same conference number represent single conference setup.
- **Call Flow** – A unique conference service setup, the logic that is used to process the conference calls. This is the process a call goes through from call setup to, to processing, to call tear down. It includes the logic, DTMF key-presses used, functions, and the recorded prompts. There are two basic call flow categories: call flows without authentication and call flows with authentication.
- **Attribute** – In terms of WYDE web services API, a data structure is used to carry attributes for call flow, DNIS and conference account (user). The attributes skeleton is defined by call flow; other attributes can only override some of them, so for instance when a user called in to the conference DNIS it gets attributes exposed by the call flow, but some of these attributes can be already altered by the DNIS. Each attribute has name, type, value, and role.
- **DNIS** – A unique set of numbers that is outputted by a phone carrier that indicates the intended destination for a particular call. It can be any length digits (although usually 10 digits). DNIS is the property of the conference account, but different DNIS numbers can be used to connect to the same conference.
- **Access Code** – A numeric code unique for DNIS that allows a host or participant or listener access to a conference call. When users call to DNIS number they being asked to enter their access code. The access code determines the conference and the user role in the conference. Different access codes can determine the same conference, for instance one access code can determine the connected user has host role, another access code can determine that connected user has participant role, and another access code can determine that connected user has listener role.
- **Host** – A user in the conference call that can make changes to the system while the conference call is in progress. Like change the security setting, change who can talk or answer, etc. Sometimes the host user is called moderator. This user role is defined in conference account. This is the most privileged role in a conference. By default, connections in this role can send and receive RTP data (i.e. the corresponding participant is allowed to speak and listen). They also are allowed to execute control actions on all connections and roles.
- **Participant** – A person in the conference who can actively participate in a call by both talking and listening. This user role is defined in conference account. Connections in this role must be allowed to send and receive RTP data by default. They can execute mute and un-mute commands on their own connections (associated with the same audio key); but not on other connections. They are allowed to drop connections within the same bundle (except where the audio key = 0).
- **Listener** – A person in the conference who can hear the conference call, but cannot speak. Their audio path is one way only (receive). This user role is defined in conference account. Connections in this role must not have the privilege to speak. They are allowed to send RTP packets to provide feedback for bandwidth adaptively on the

stream sent by the bridge. They are allowed to drop connections that are within the same bundle (except where the audio key = 0). Note: users in listener role can be unmuted to enable them to talk; however, the listener group as a whole will never be unmuted.

## **Chapter 2: Step-by-step WYDE Voice Demo Software Virtual Machine Configuration**

This chapter will explain how to configure the WYDE Voice Software that was provided as VMware Image for demonstration purposes.

### ***Step 1: Get VMware WYDE Voice Software Demonstration Image***

VMware image of the fully installed system with the WYDE Voice conferencing bridge software can be downloaded from Internet using your FTP username and password. For instance, your URL could be the following:

```
ftp://YourFtpUsername:YourFtpPassword$123-  
@ftp.wydevoice.com/vmware/WYDE-2.1.32.zip
```

Please download the archive file and unzip it into folder on your hard drive.

This sample shows download URL of the WYDE Voice conferencing bridge software version 2.1.32 for CentOS 5 VMware image. The actual version of the WYDE software may be different for you.

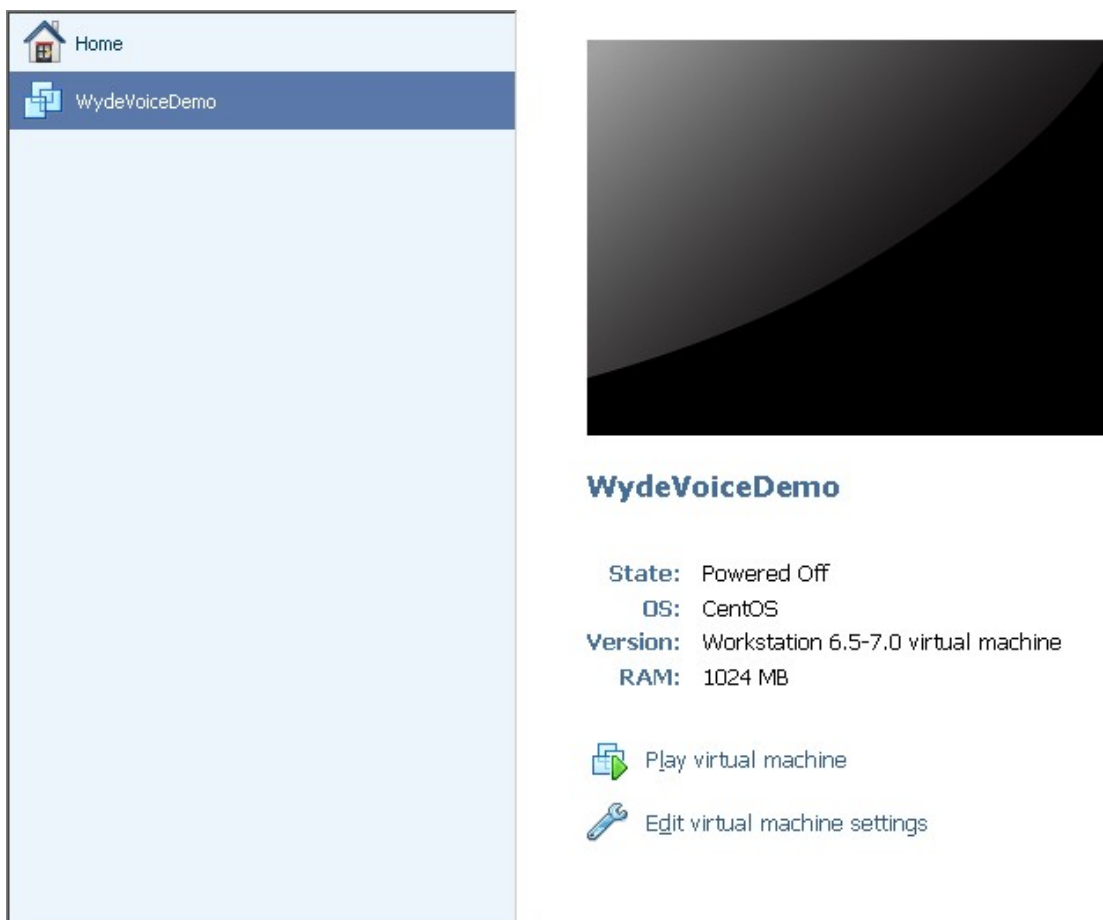
### ***Step 2: Open and Start VMware WYDE Voice Software Demonstration Image***

Please run VMware Player. You will see the screen similar to shown on Figure 1.



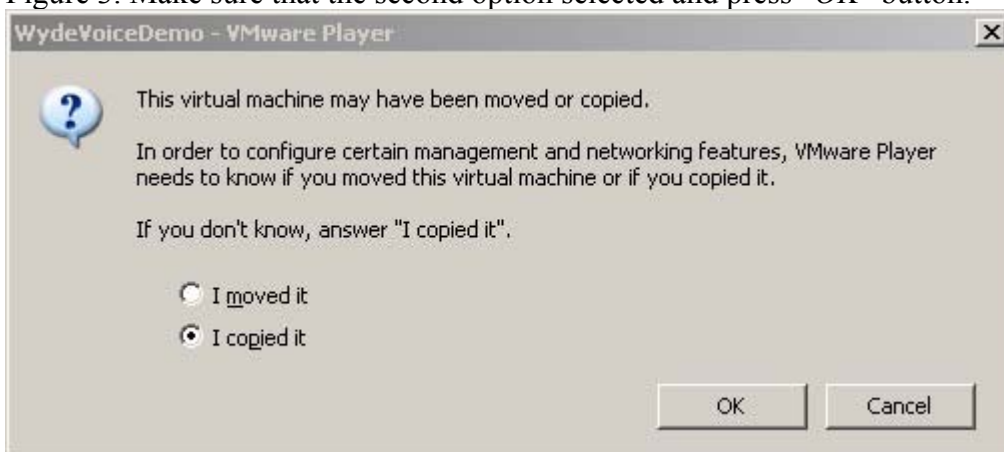
**Figure 1: VMware Player Welcome Screen**

Press “*Open a Virtual Machine*” link button and point to the unzipped received image on your hard drive; actually you should point to *WYDEVoiceDemo.vmx* file. After you did it you will see the screen similar to shown on Figure 2.



**Figure 2: Open VMware Virtual Machine**

Press “*Play virtual machine*” button to run the machine. Because this virtual machine was created on another computer you may receive the warning message similar to shown on Figure 3. Make sure that the second option selected and press “OK” button.



**Figure 3: VMware Virtual Machine “moved or copied” Warning**

When the virtual machine started you will see the screen similar to shown on Figure 4.

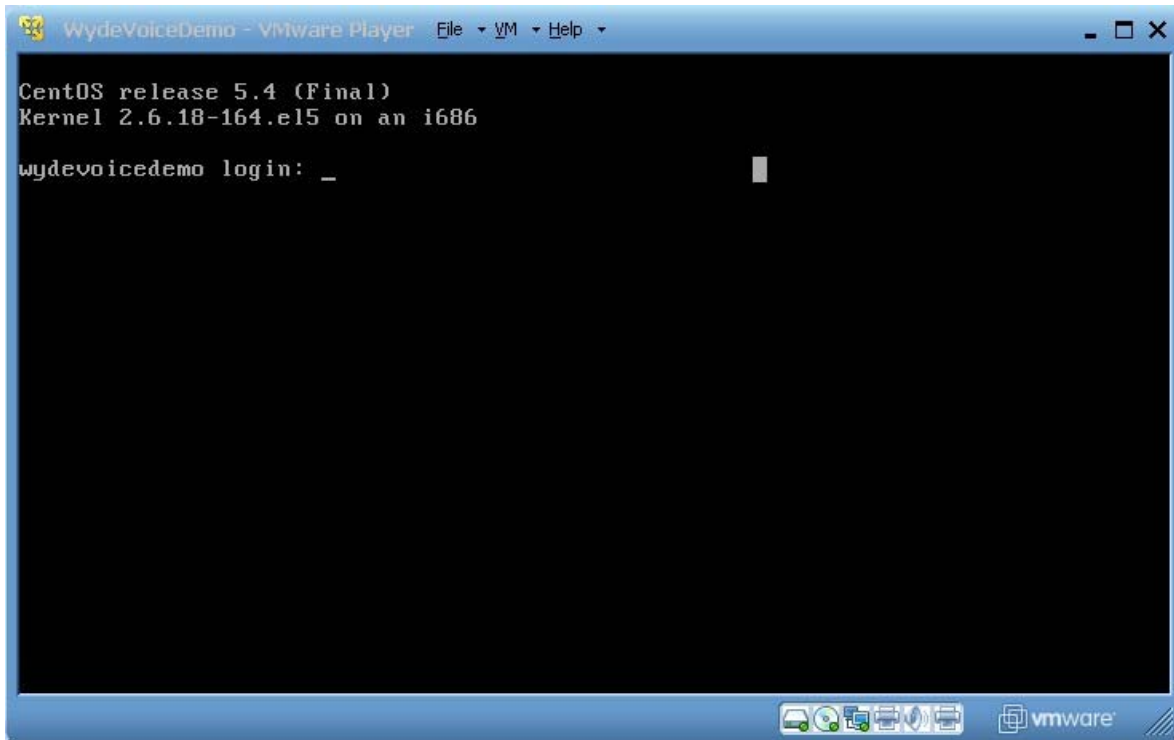


Figure 4: VMware Virtual Machine Login Screen

### ***Step 3: Login and Configure VMware WYDE Voice Software Virtual Machine***

To login into system you should use the following Linux credentials:

- Login: *root*
- Password: *wydevoice*

Note once you know the IP of your virtual machine in addition to VMware Player you can use any SSH program to connect and login into the machine.

After you login to the image you will see short memo what should be done to make initial configuration of WYDE Voice Conference Server, as shown on Figure 5.

```

Kernel 2.6.18-164.el5 on an i686

wydevoicedemo login: root
Password:
Last login: Wed Feb 24 17:30:07 from 192.168.206.1
*****
Welcome to WYDE VOICE Conference server
*****
To start you need to configure the bridge to use your new IP address
1. Find out what is your current IP address on this host (ifconfig)
2. Run commands:
   "service postgresql start" (if it is not running yet)
   "wyde ip-set ip <address>"
   "service wyde restart" (it is important that postgres is restarted)

After that you can try to place SIP calls to sip:100<address>:5060
You can enter any access code when prompted. All calls with the
same access code will join into the same confence

Good luck.
WYDE VOICE dev team.
support@wydevoice.com

*****
[root@wydevoicedemo ~]# _

```

Figure 5: WYDE Voice Software Login Memo

Please run `ifconfig` command to find out what is your current IP address of this virtual machine. For example Figure 6 shows that `192.168.213.128` is the current IP of the virtual machine.

```

[root@wydevoicedemo ~]# ifconfig
eth0      Link encap:Ethernet  HWaddr 00:0C:29:6F:C0:13
          inet addr:192.168.213.128  Bcast:192.168.213.255  Mask:255.255.255.0
          inet6 addr: fe80::20c:29ff:fe6f:c013/64 Scope:Link
          UP BROADCAST RUNNING MULTICAST  MTU:1500  Metric:1
          RX packets:169 errors:0 dropped:0 overruns:0 frame:0
          TX packets:216 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:1000
          RX bytes:17239 (16.8 KiB)  TX bytes:28845 (28.1 KiB)
          Interrupt:67 Base address:0x2000

lo        Link encap:Local Loopback
          inet addr:127.0.0.1  Mask:255.0.0.0
          inet6 addr: ::1/128 Scope:Host
          UP LOOPBACK RUNNING  MTU:16436  Metric:1
          RX packets:8 errors:0 dropped:0 overruns:0 frame:0
          TX packets:8 errors:0 dropped:0 overruns:0 carrier:0
          collisions:0 txqueuelen:0
          RX bytes:560 (560.0 b)  TX bytes:560 (560.0 b)

[root@wydevoicedemo ~]# _

```

Figure 6: Find out the Current IP Address

Next you should start *PostgreSQL* (if it is not running yet) using the command:  
`service postgresql start`

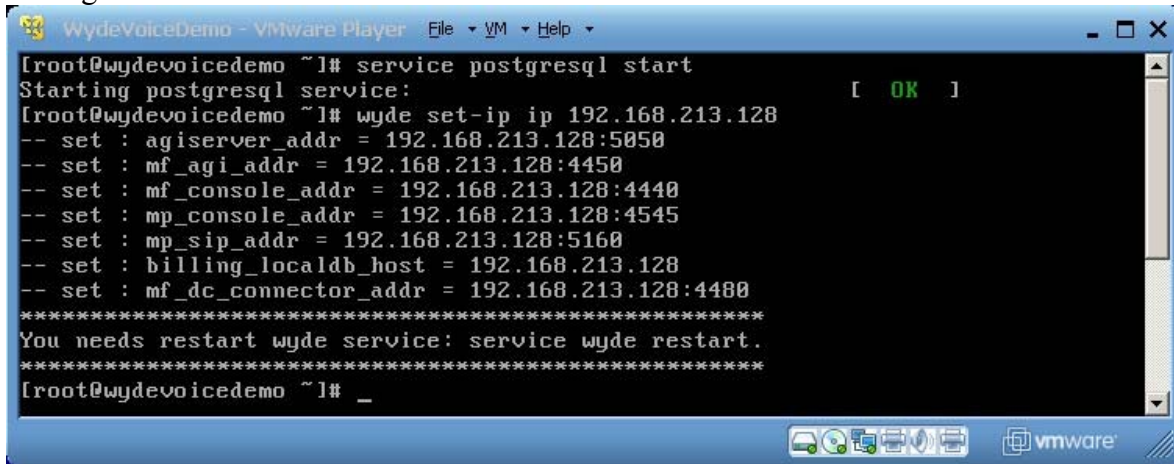
After that you should configure the WYDE Voice software to your IP address using the command:

```
wyde set-ip ip <your IP address>
```

in this sample the following command should be run:

```
wyde set-ip ip 192.168.213.128
```

see Figure 7 for details.



```
[root@wydevoicedemo ~]# service postgresql start
Starting postgresql service: [ OK ]
[root@wydevoicedemo ~]# wyde set-ip ip 192.168.213.128
-- set : agiserver_addr = 192.168.213.128:5050
-- set : mf_agi_addr = 192.168.213.128:4450
-- set : mf_console_addr = 192.168.213.128:4440
-- set : mp_console_addr = 192.168.213.128:4545
-- set : mp_sip_addr = 192.168.213.128:5160
-- set : billing_localdb_host = 192.168.213.128
-- set : mf_dc_connector_addr = 192.168.213.128:4480
*****
You needs restart wyde service: service wyde restart.
*****
[root@wydevoicedemo ~]# _
```

Figure 7: Update WYDE Voice Software IP Address

After that you must restart WYDE services using the command:

```
service wyde restart
```

At this moment the WYDE voice software on this virtual machine is fully configured. Now you can make test calls to the bridge, use Web Administration Interface and Web Control Console as it will be described in the next chapter of this guide.



## Chapter 3: Trial of WYDE Voice Demo Software

At this moment we assume that you have completed the configuration of the WYDE Voice Software that was provided as VMware Image as it was described in Chapter 2: Step-by-step WYDE Voice Demo Software Virtual Machine Configuration.

This chapter will explain how you can make test calls to the bridge, how to run Web Administration Interface and Web Control Console.

### *SIP Calls to the Bridge*

If the configuration was successful you should be able to call on your bridge now. You can use [X-Lite](#) or [SJphone](#) softphones to make VOIP call on your bridge using the IP address that you found out during installation (see Figure 6) and that was used to configure your WYDE bridge (see Figure 7).

If you use X-Lite software you should configure it as shown on Figure 8. Run X-Lite, make right mouse click on your program, select “*SIP Account Settings...*”, press “*Add...*” button and enter account properties as shown on this figure.

The screenshot shows the 'Properties of Account 1' dialog box in X-Lite. The 'Account' tab is active. The 'User Details' section includes fields for Display Name (Administrator), User name (admin), Password (masked with dots), Authorization user name (admin), and Domain (192.168.213.128). The 'Domain Proxy' section has a checkbox for 'Register with domain and receive incoming calls' which is unchecked. Below it, 'Send outbound via:' has two radio buttons: 'domain' (unchecked) and 'proxy' (checked). The 'Address' field next to 'proxy' contains '192.168.213.128:5060'. At the bottom, the 'Dialing plan' field contains '#1\|a\|a.T;match=1;prestrip=2;'. The dialog has 'OK', 'Cancel', and 'Apply' buttons at the bottom right.

Figure 8: X-Lite SIP Account Settings

Your user name and authorization user name should be *admin*; the password is *admin* as well. In addition you should enter *<your IP address>* into domain field and select proxy and enter *<your IP address>:5060* into proxy address field. After that you will be able to make SIP calls the bridge to the IP address that you used during configuration. For instance you can call to number 10 as shown on Figure 9.

If you are using SJphone software you do not need any additional configuration. You can just make the call to *10@<your IP address>:5060* (*10@192.168.213.128* in our sample) as shown on Figure 10.



Figure 9: X-Lite SIP Call to the Bridge

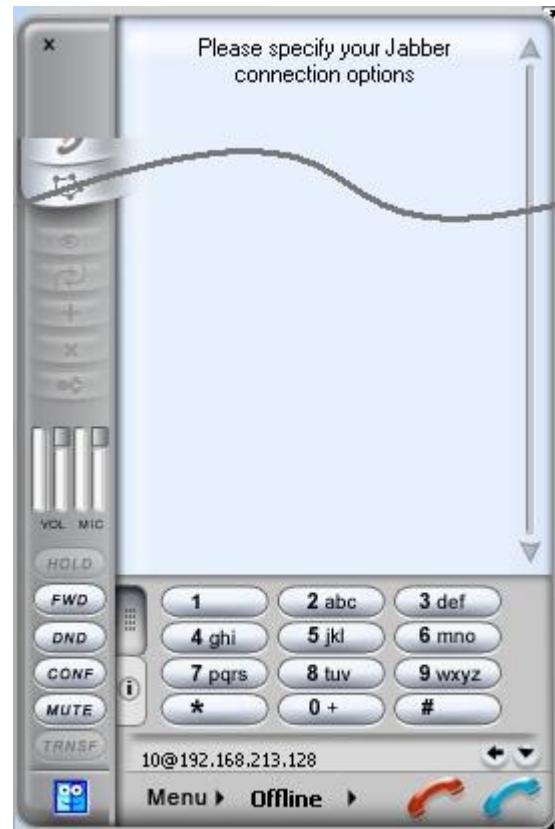


Figure 10: SJphone SIP Call to the Bridge

When you call to number *10* you can enter any access code when prompted. All calls with the same access code will join into the same conference.

### ***Web Administration Interface***

In addition you should be able to open Web Administration Interface in your browser from this virtual computer. You should use the IP address of this computer to do that, i.e. you should type *http://<your IP address>/* (*http://192.168.213.128/* in our sample) in your browser URL bar. You will see a screen similar to the shown on Figure 11. Use PIN: *admin* and password: *admin* to login to the system.



WYDE VOICE

24/7 Professional Customer Service  
TOLL FREE 866.508.9020

CONFERENCE APPLIANCE MANAGER

PIN:

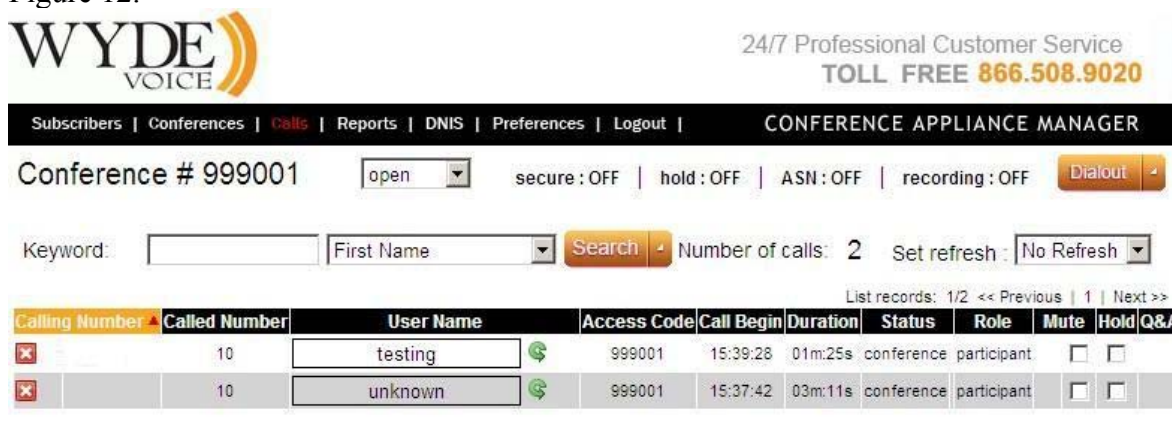
Password:

LOGIN

Problems logging in? Contact Support at [support@wydevoice.com](mailto:support@wydevoice.com)

**Figure 11: Initial Web Administration Login Screen**

When the calls are made on the bridge you should be able to see the conferences and the calls using Web Administration Interface; you will see the screen similar to shown on Figure 12.



WYDE VOICE

24/7 Professional Customer Service  
TOLL FREE 866.508.9020

Subscribers | Conferences | **Calls** | Reports | DNIS | Preferences | Logout | CONFERENCE APPLIANCE MANAGER

Conference # 999001  secure : OFF | hold : OFF | ASN : OFF | recording : OFF

Keyword:  First Name   Number of calls: 2 Set refresh:

List records: 1/2 << Previous | 1 | Next >>

Calling Number	Called Number	User Name	Access Code	Call Begin	Duration	Status	Role	Mute	Hold	Q&A
<input type="checkbox"/>	10	testing	999001	15:39:28	01m:25s	conference	participant	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/>	10	unknown	999001	15:37:42	03m:11s	conference	participant	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

**Figure 12: Conference Calls Screen when the First Two Calls made on the Bridge**

### ***Web Control Console***

Additionally you are able to use Web Control Console (WCC) from this virtual computer. You should use the IP address of this virtual computer, i.e. you should type *http://<your IP address>/wydevoice/* (for instance *http://192.168.213.128/wydevoice/*) in your browser URL bar. You will see a screen similar to the shown on Figure 13.

**WYDE VOICE**

**Account Sign Up**

First Name:

Last Name:

Email:

Password:

☐ I agree to the [Terms and Conditions](#)

**Account Login**

Email:

Password:

☐ Remember me on this computer.

[Forgot Password?](#)

**FAQs**

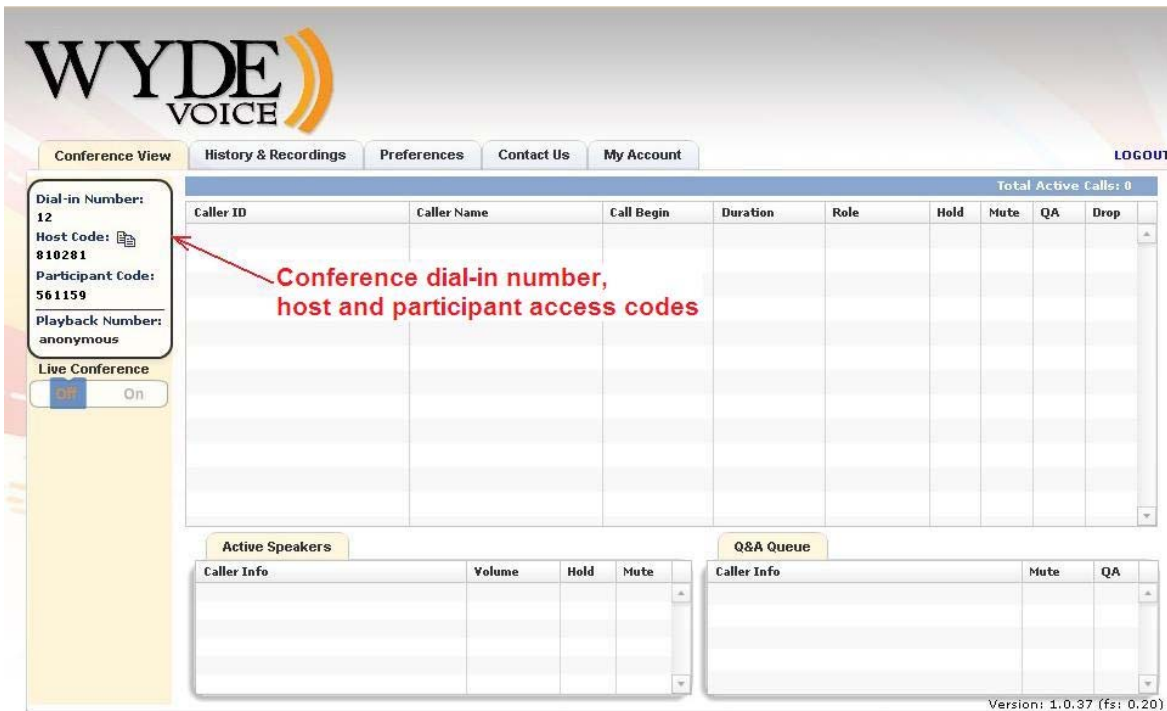
[What is this?](#)  
[How does it work?](#)  
[What is Conference View?](#)  
[What are Preferences?](#)  
[Can I set conference preferences during my live conference calls?](#)  
[What are host web controls?](#)  
[When should I use host web controls?](#)  
[How do I get a Free Conferencing account with host web controls?](#)  
[What is Q&A queue and how does it work?](#)  
[How many callers can be on a single conference call?](#)  
[What are the costs involved in a Wyde Voice account?](#)

[What is Conference History and Recording?](#)  
[What is Active Speaker?](#)  
[Is there a limit on the length of a Free Conferencing conference call?](#)  
[Is there a limit to how many conference calls I can make?](#)  
[I'm having problems creating an account?](#)  
[I'm having problems connecting to the web interface?](#)  
[Do I get a report of my conference call?](#)  
[Can I record my conference calls?](#)  
[How do I access my recorded conference calls?](#)  
[Is this a VoIP service or an Internet service using my computer?](#)  
[Can international callers participate?](#)

**Figure 13: Initial Web Control Console Login Screen**

To use Web Control Console you should sign up your account first – you should enter your first and last name, email and password. Your email and password are being used as your credentials to login into Web Control Console.

When you signed up and login into console you will see dial-in number, host and participant access codes to call the conference on the bridge, see Figure 14 for details. The virtual WYDE Voice demo server provided with VMware image is configured to make the calls to number 12 (call flow: SPECTEL), but you may configure the software to another number.



**Figure 14: Conference Dial-in Number, Host and Participant Access Codes**

If you would like to configure WYDE Web Control Console to use another dial-in conference number to call and/or you would like to configure playback conference number you should perform the following configuration changes:

- i) Using Web Administration Interface register conference DNIS number (call flow: SPECTEL); in this call flow set “*Realtime protocol*” attribute value equal “on”, “*Callers database*” attribute value equal “on”, and “*Playback number*” attribute value equal to playback conference DNIS number;
- ii) Using Web Administration Interface register playback conference DNIS number (call flow: PLAYBACK) and in this call flow set “*Reffered DNIS number*” attribute value equal to the DNIS number registered in the previous step;
- iii) Update the configuration file *supportmodule.properties* in folder: */opt/tomcat/webapps/supportmodule/WEB-INF/classes*

The following parameters should be changed:

conferencePN – should be equal to number registered in the step i);

playbackPN – should be equal to number registered in the step ii).

After this configuration has made you should restart *tomcat* service using the command:  
`service tomcat restart`

Please read “Web Administration Interface – User Guide”, Chapter 2: Web Administration Interface, Section: “DNIS Management” if you need assistance in DNIS creation/modification and call flow attributes editing.

Instead of Web Administration Interface you can use command line interface to create new DNISes and modify their attributes. To create new DNIS you should use the command:



```
wyde did-add number <DNIS (dial-in) number> callflow <call  
flow name> [description <DNIS description>]
```

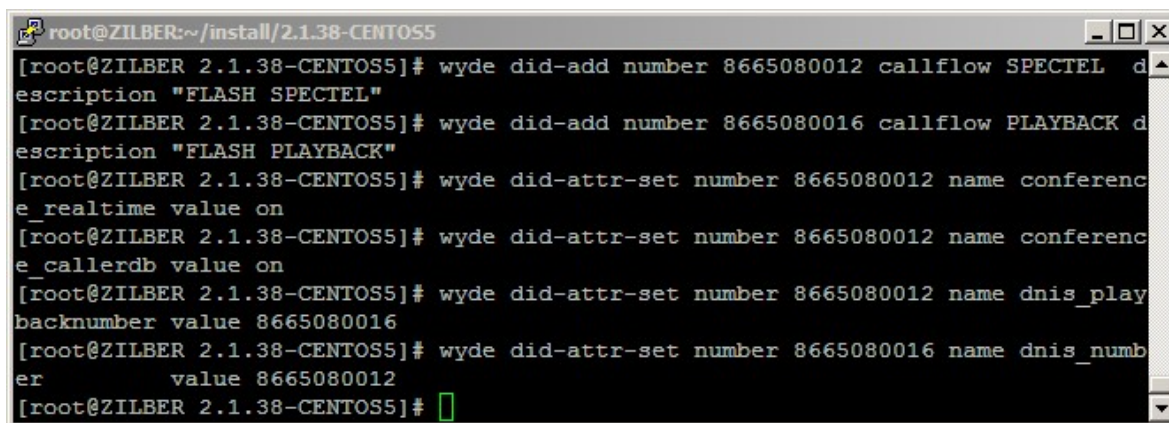
To update DNIS attribute value you should use the command:

```
wyde did-attr-set number <DNIS (DID) number> name <attribute  
name> value <attribute value>
```

For instance if you would like that WYDE Web Control Console use DNIS *8665080012* for conference calls and DNIS *8665080016* for playback you can run the following commands:

```
wyde did-add number 8665080012 callflow SPECTEL description "FLASH SPECTEL"  
wyde did-add number 8665080016 callflow PLAYBACK description "FLASH PLAYBACK"  
  
wyde did-attr-set number 8665080012 name conference_realtime value on  
wyde did-attr-set number 8665080012 name conference_callerdb value on  
wyde did-attr-set number 8665080012 name dnis_playbacknumber value 8665080016  
wyde did-attr-set number 8665080016 name dnis_number value 8665080012
```

See Figure 15 for details.

A screenshot of a terminal window titled 'root@ZILBER:~/install/2.1.38-CENTOS5'. The terminal shows a series of commands being entered and executed. The commands are: 'wyde did-add number 8665080012 callflow SPECTEL description "FLASH SPECTEL"', 'wyde did-add number 8665080016 callflow PLAYBACK description "FLASH PLAYBACK"', 'wyde did-attr-set number 8665080012 name conference\_realtime value on', 'wyde did-attr-set number 8665080012 name conference\_callerdb value on', 'wyde did-attr-set number 8665080012 name dnis\_playbacknumber value 8665080016', and 'wyde did-attr-set number 8665080016 name dnis\_number value 8665080012'. The prompt '[root@ZILBER 2.1.38-CENTOS5]#' is visible at the start of each line. The terminal window has standard Linux window controls (minimize, maximize, close) in the top right corner.

```
root@ZILBER:~/install/2.1.38-CENTOS5  
[root@ZILBER 2.1.38-CENTOS5]# wyde did-add number 8665080012 callflow SPECTEL d  
escription "FLASH SPECTEL"  
[root@ZILBER 2.1.38-CENTOS5]# wyde did-add number 8665080016 callflow PLAYBACK d  
escription "FLASH PLAYBACK"  
[root@ZILBER 2.1.38-CENTOS5]# wyde did-attr-set number 8665080012 name conferenc  
e_realtime value on  
[root@ZILBER 2.1.38-CENTOS5]# wyde did-attr-set number 8665080012 name conferenc  
e_callerdb value on  
[root@ZILBER 2.1.38-CENTOS5]# wyde did-attr-set number 8665080012 name dnis_play  
backnumber value 8665080016  
[root@ZILBER 2.1.38-CENTOS5]# wyde did-attr-set number 8665080016 name dnis_numb  
er value 8665080012  
[root@ZILBER 2.1.38-CENTOS5]#
```

Figure 15: WCC DNIS Creation and their Attributes Modification

After these changes are made and *tomcat* service is restarted you are able to call to new number and use playback number that you have configured.

## **Appendix A: Support Resources**

If you have difficulty with this guide and any of the procedures listed herein, please contact us using the following support resources.

### ***Support Documentation***

In addition to this Guide, you may obtain other WYDE Voice documentation from WYDE Voice or from the support section of <http://www.wydevoice.com/>.

### ***Web Support***

Our support website is available 24 hours a day, 7 days a week, and 365 days a year at <http://www.wydevoice.com>. You may download patches, support documentation and other technical support information.

### ***Telephone Support***

For difficulties with any procedures described in this Guide, please contact us at 866-508-9020 during our normal phone support hours of 7:00 am to 6:00 pm Pacific Standard Time (PST). An engineer will respond to your inquiry within 24 hours.

### ***Email Support***

You may also email us your questions at [support@wydevoice.com](mailto:support@wydevoice.com). We will respond to your question within 24 hours.