Guide to Buying An

Audio Conferencing Bridge

WYDE Voice LLC White Paper

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Executive Summary

In this white paper report, we highlight some of the questions you should ask if you're shopping for a new audio conferencing bridge. With the advent of software based conferencing bridges running on the latest most powerful hardware, we've entered a bold new era of audio conferencing. If you still have a legacy bridge (like Spectel) or your new unified communications PBX doesn't have sufficient conferencing capacity, one of the new SIP compliant conferencing bridges might be the perfect fit. The big question is which one you should choose?

There are several solid new audio conferencing bridges on the market. To keep it simple, we've used the WYDE Voice SB-HD Conferencing Bridge as an example of the new generation conferencing bridge, but the same questions could be applied to any new platform.

Question 1--One of the most frustrating things about getting introduced to new technology is learning how to use it. Ideally, the new bridge should be able to emulate the call flow of your old bridge. Does the bridge that you are evaluating allow call hosts and participants to interact and control the new bridge in the same way as the old bridge—the same call flow—e.g. DTMF commands for muting, unmuting, recording, and recording playback, etc?

WYDE Response: Flexible call-flow design. There are several main types of call flows available on the WYDE bridges:

- **CONF**: The CONF call is a normal conference call setup with no security. All participants use the conference number as the access code and the only differentiator between host and participant is the termination of the login – hosts use *, participants use #.
- **SPECTEL**: This call flow is similar to the CONF call flow, except the host and participants use different access codes. Also, you can give each participant their own access code, have them share an access code, or give groups of participants their own access codes.
- **Event Conferencing:** The Event Conferencing call flow is a specialized call flow that allows multiple panelists to have a discussion while many more listeners listen to the conversation. For more information, refer to http://www.simpleevent.com/faq.asp
- **OPERATOR**: This call flow is used for administrative purposes as quick technical support; it is not used for actual conferences, but it is used to assist the users in case of any problems. The operator can monitor the conferences, connect to different conferences, receive requests from the users, attach a user to a different conference, dial out, etc. When an operator connects to her operator conference account, she is immediately notified if new user requests have been assigned to her queue. Alternatively, she can go into conference surveillance and management mode. Operator functions can be initiated using touch tone commands or from the web interface (Web Administration Interface and Flash Operator Console).
- **PLAYBACK**: This call flow is used to playback previously recorded conferences. The user calls the PLAYBACK call number, enters the conference access code, and is then able to playback the recorded conference.
- **Custom**: WYDE bridge owners also have the option of using the web services API to develop completely custom call flows for their company.

Finally, the bridge owner can deploy one or a combination of the above call flows based on DNIS or conference account.

Question 2--Most desktop computer users are accustomed to having a simple Graphical User interface (GUI) to control most of the applications they use every day. Does the bridge you are considering have a good GUI for call hosts, operators, and system administrators?

<u>WYDE Response</u>. The bridge has several user interface features that make it easy for operators, call hosts, and regular participants to control the conferencing bridge.

A. <u>Adobe Flash-based Dashboard for conference call hosts</u>, also known as a *Moderator Console* allows a host to Mute/Drop/Place on Hold Any Guest; lock a conference and prevent unauthorized people from joining the call; easily create/manage real-time question and answer queues. Graphical Control of Call Recording makes recording of conference calls easy.

See illustration below.
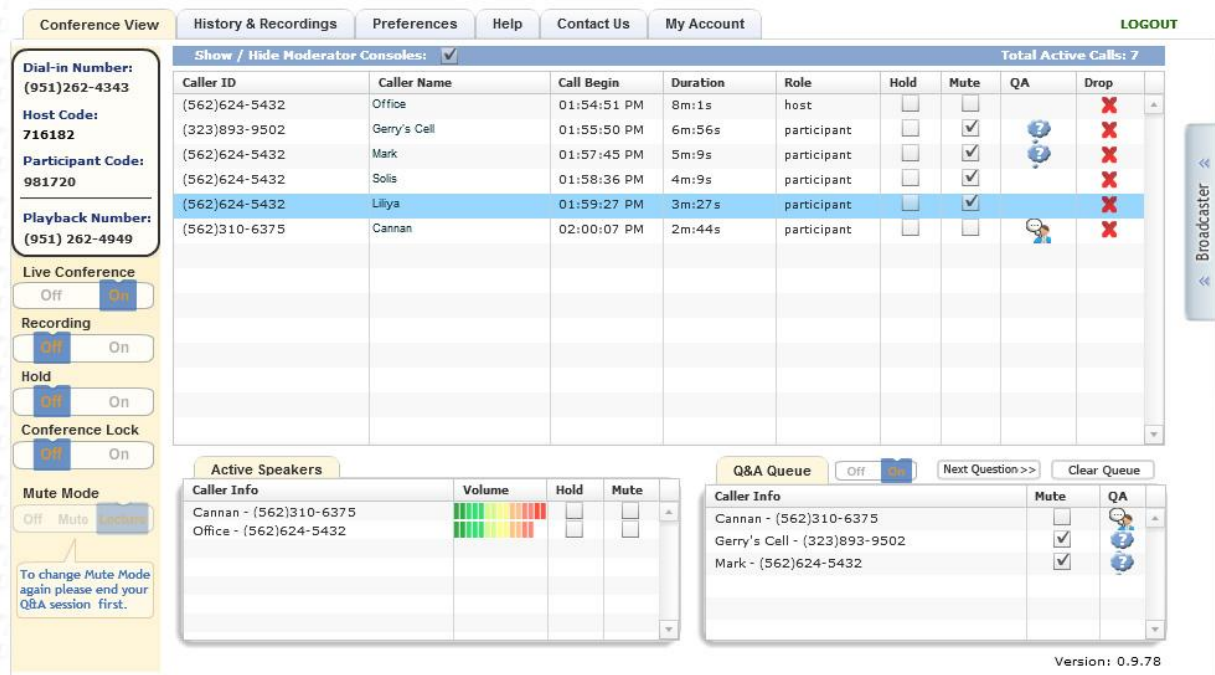
The interface also provides the following functionality

- Real-Time Broadcast of Any Audio Files during the Conference Call to leverage previously recorded materials for multiple calls.
- Individually record and color code index each caller line connecting to a conference so that individuals playing back recordings of the conference call can isolate and playback specific callers who joined the conference call.

B. <u>Adobe Flash-based Dashboard for Operators</u>, also known as the Operator Console allows operators to provide call monitoring of any conference on the bridge, see real-time DTMF commands being entered by guest callers, transfer callers to the correct conference, dial-out

from the bridge to add callers to a conference, and assist in Q&A session management. See illustration below.

C. Adobe Flash-based Dashboard for Administrators, also known as the Administrator Console, (in addition to functions provided to operators) also provides comprehensive reports allowing an Administrator to verify which calls in which conferences took place. These reports are available through the standard Web Administration interface that comes with the bridge. They include the following reports:
   o Calls report
      ✓ gives Administrator the possibility to review individual CDRs
   o Conference Report
      ✓ allows an Administrator to view a list of conferences which have taken place in the past and this report identifies previously recorded conferences which can be played back as required. In addition the recorded conference is illustrated in the form of a Gantt chart identifying individual speakers taking part in the conference.
   o DNIS report
      ✓ Lists call distribution by different called numbers.
   o Disconnect report
      ✓ Lists call distribution by different disconnect reasons. This is very helpful for troubleshooting.
   o Load charts
      ✓ Shows actual port utilization over the desired time interval.

Question 3--Active Speaker Notification.   Does your new bridge tell you who is talking during a conference call, and which lines are generating the noise that is impacting your conference call?

WYDE Response: The WYDE bridge delivers extremely accurate active speaker notification data to the flash GUI, refreshing every 100 milliseconds (Ten times per second). Based on the active speaker info, the host can selectively mute participants and block noisy lines.

Question 4--There are now over 500 Million Skype users worldwide. Is the new bridge capable of receiving free calls from soft phones like Skype?

WYDE Response: The bridge can be assigned multiple Skype addresses and can receive calls from Skype and many other soft phones. To see a live service that has enabled this feature, check out http://freeconferencecallhd.com/skypeinstructions.html

Question 5--Does the bridge support High Definition (HD) voice codecs?

WYDE Response: In addition to standard low frequency voice codecs like G.711 and G.729, the WYDE bridge supports the following HD voice codecs: G.722, G.722.1, G.722.2, and ISAC. Why is HD Voice important? Traditional (PSTN) calls deliver sounds representing only a fraction of the range one would hear in a live conversation — the rest is deliberately filtered out. HD voice, while still not transmitting all audible frequencies, delivers enough of the ones relevant to human speech such that many listeners say it seems like the speaker is in the same room. Long calls, especially conference calls with multiple speakers with different accents, become less tiring and thus more productive.

Question 6--Does the new bridge integrate easily with your preferred web conferencing solution?

WYDE Response: The WYDE Voice conferencing software provides a Web Services API that allows a web conferencing app to query and manage conferences and calls occurring on the bridge, get information about live conferences and calls, drop specific conferences and calls, place conferences and callers on hold, mute/unmute conferences and callers, make the conference secure, engage/disengage Q&A sessions, start/stop conference recording, dial out to another users, etc., configure subscribers and their conference accounts including access code used for local authentication, maintain DNIS and call flow management. Additionally this API helps to get information about live calls in real time as well as gather information on completed calls. In summary, the WYDE bridge web services API makes it easy to integrate the bridge with 3rd party web conferencing applications like Adobe Connect, Persony, WebEx, FreeSee and others.

Question 7--Does the new bridge integrate easily with external billing applications?

WYDE Response: The WYDE bridge software can store and transmit CDRs (Call Detail Records), which provide comprehensive data about completed conferences and calls. CDRs can be delivered to an external billing system in one of the following ways:

- The External billing system pulls new CDRs from the internal database regularly at predefined intervals. In this case you would use data created by the standard *localdb* billing adapter in a *dnca_calls* database. You can write your own routine, which in a given time interval will periodically take new CDR data from the WYDE bridge billing database and place them into your own database that you use for your billing procedures.
- Using the custom Billing Adapter, the bridge can push CDRs to an external billing system on the fly in the real-time. Usual push scenarios are:
    - The Billing Adapter inserts CDRs into an external database. Such custom billing adapters could store CDR information in your own database in the manner that you prefer.
    - The Billing Adapter creates CSV files and makes them available on the FTP; the remote system takes them from FTP itself.
    - The Billing Adapter sends CDRs to the external system using proprietary protocol (TCP based, UDP based, SOAP, etc).
- CDR information could be sent to your FTP using a special script; this script is usually being run daily. It takes CDR records for the last day, creates a file with this data in the requested format and sends this file to the specified FTP.

Question 8--What kind of integration with authorization systems does the bridge provide?

WYDE Response*: Authorization Integration* for conference authorization, i.e. defining the right to connect to the conference and the specific roles (host/moderator/listener) in the conference can be made in one of the following ways:
- You can turn off authorization completely, i.e. anyone who called to the conference DNIS number is allowed to connect to the conference regardless of access code entered. Usually this approach is used in CONF call flow.
- Authorization can be made via a local database such that when a person dials the conference DNIS number, they are then asked to enter an access code. This access code is then verified in the local *dnca* database against a subscribers' conference account definitions. User roles in the conference (i.e. host, participant, listener roles) are assigned according to the DNIS number and access code used. This approach is usually used in the SPECTEL call flow.
- Authorization can be made via a RADIUS server using the WYDE data dictionary by way of the WYDERadius standard authorization adapter.
- Authorization can be made via LDAP using the WYDE data dictionary by way of the WYDELdap standard authorization adapter. In this case authorization of callers in conferences is being made based on the Active Directory security data of your organization.
- A custom authorization adapter can be written to determine which users can connect to the conference and what role that user will be granted, e.g. user, host, participant, or listener.

Typically, the information can be taken from your external SQL database (for instance using RADIUS server) or from Active Directory Domain Controller or others.

Question 9--Is the system easy to deploy?

WYDE Response: WYDE Voice conference bridges are designed on the "Datacenter in a box" concept, which means that they can be deployed quickly with minimal technical resource. The platform is virtually "plug and play".

"Datacenter in a box" deployment consists of installing the specified conference hardware which is then configured for use in that conferencing telephony communications solution. WYDE conference bridges are designed to be attached to your network in appliance-fashion. They are, for all intents and purposes, servers that can be installed and managed as such.

WYDE bridges are typically deployed in one of two configurations:
- "*Small*" – the equipment can support up to 2,000 concurrent ports of Audio Conferencing functionality, i.e. can support up to 2,000 simultaneous calls;
- "*Medium*" – the equipment can support up to 10,000 concurrent ports of Audio Conferencing functionality, i.e. can support up to 10,000 simultaneous calls.

Question 10--Many legacy bridge owners have a difficult time finding spare parts for their discontinued bridges, and if they are able to find parts, they pay a fortune. Does the bridge you are considering use off-the shelf commodity hardware that takes advantage of the latest processors and server architecture, and have a solid technical team providing around-the-clock support?

WYDE Response: WYDE's software based bridge runs on off-the-shelf server hardware from Dell. As the second largest server manufacturer in the world, Dell has developed an excellent hardware support network. WYDE provides 24X7 software application support via phone and web from centers in Long Beach, CA; Washington, DC; and Moscow.

Question 11--Does the new bridge help you to save on international and domestic telephone costs? For example many bridges require international participants to make expensive international calls to connect to an audio conference?

WYDE Response: Bridges can be deployed to allow 'Geo-Dispersed conferencing', allowing several bridges to be deployed in different regions or countries yet acting as one large conference bridge. This means that calls are being made to separate bridges located in various regions, but these calls are connected to the same conference. A brief description of this feature of the WYDE bridge can be found below
- Bridges communicate with each other to build a bigger virtual bridge.

- Users can call any bridge on the geo-dispersed network and be connected into the same conference call.
- Connection is very efficient – only one SIP/RTP channel required per conference to join two bridges.
- Each WYDE bridge dynamically adjusts to network problems between geo-dispersed systems.
- All group operations (mute, recording, etc) and UI can be utilized from any bridge.
- Up to 16 bridges can be connected together.
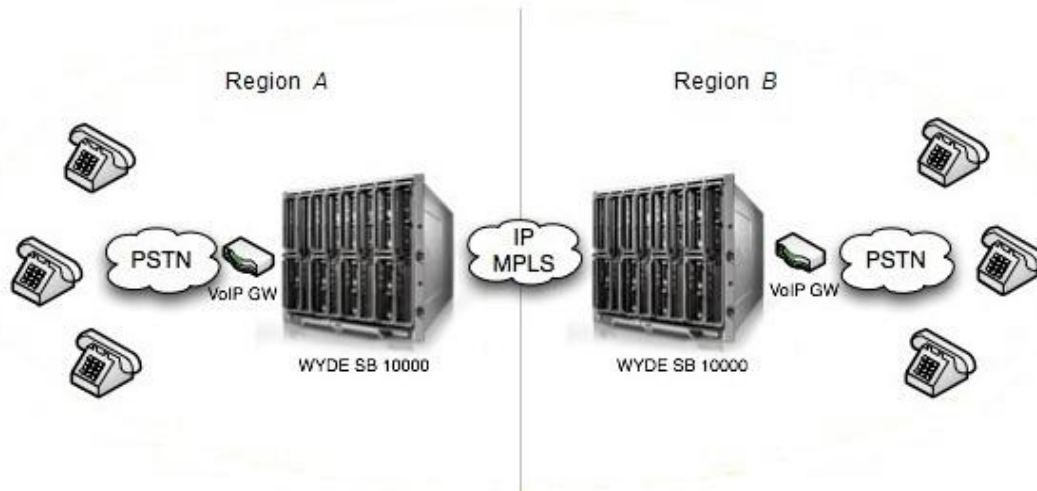
Question 12--Is the system capable of connecting large numbers of people calling in simultaneously at the top and bottom of the hour?

WYDE Response: The bridges can be deployed in any number of different configurations, with varying numbers of media servers and front end modules depending on specific customer requirements. You simply add more low-cost front-ends to handle the extra lobby traffic.

Question 13--Does your new bridge provide sophisticated diagnostic and monitoring tools for maintaining its health and minimizing downtime?

WYDE Response: The WYDE Voice conference equipment is under constant SNMP monitoring. All counters and MIBs are integrated into Open NMS which provides detailed reports about past and current problems.

Question 14--Does the system support both TDM and SIP voice traffic?
WYDE Response: The bridge is natively SIP based so it connects to Soft switches and SIP compliant PBXs via Ethernet. If a customer needs to accommodate TDM traffic from a legacy switch or PBX, they simply deploy a commodity VoIP gateway as shown in the diagram below. The VoIP gateway converts the TDM signal into a SIP signal. Both Cisco and Audio Codes make quality VoIP gateways

Question 15--If you've moved to a VoIP telephony network, is your conferencing bridge able to adjust to congestion over the network and maintain conference call voice quality?

WYDE Response: The WYDE bridge implements advanced jitter buffer management and packet loss concealment (NetEQ) to enhance voice traffic delivery when there is network congestion. Jitter buffer management is used to ensure a continuous and smooth play-out of audio transmitted over the VoIP network. Additionally the WYDE bridge supports some codecs (like SILK) that have low bite rate redundancy (LBRR), also called in-band forward error correction (FEC), which makes them robust and resistant to packet loss.

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Summary:

In the white paper above, we highlighted some of the questions you should ask if you're shopping for a new audio conferencing bridge. With the advent of software based conferencing bridges running on the latest most powerful hardware, we've entered a bold new era of audio conferencing. Many features that can improve the conferencing experience, increase reliability, and reduce costs can be found in new platforms like the WYDE Voice SB-HD1000/10,000.

For more information, please visit our Web site at www.WYDEvoice.com or send an email to sales@WYDEvoice.com.

WYDE Voice LLC,   110 W. Ocean Blvd, Suite D, Long Beach, CA 90802

About WYDE Voice

WYDE Voice develops and manufactures the most advanced audio conferencing bridges on the planet. We scale higher (up to 10,000 concurrent users), support the widest range of HD voice codecs, integrate easily with legacy billing and authentication infrastructure, offer the most intelligent call host conference control panel on the market, and have created the first network of conferencing bridges that talk to each other seamlessly. We are based in Long Beach, CA and won the VON Magazine "Best in Show" award upon our launch in 2008.