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Digital Speech Level Analyser

Network Voice Services User Guide

Revision 1.3

Malden Electronics Ltd

Table of Contents

End User License Agreement **2**

Introduction **3**

What is a DSLA Service? 3
What is NVS? 3
Multiple NVS Installations 4

Getting Started **5**

System Requirements 5
Installation 5
NVS User Interface 6
Quick-Start 7
Using your Playlists and Schedules 9

How To **12**

How to make a H323 (NetMeeting) call 12
How to make Cisco Call Manager routed calls 14
How to make H323 gatekeeper routed calls 16
How to call MS Messenger (SIP) 18
How to make SIP calls with authentication 20

Frequently Asked Questions **22**

Connecting Issues 22
Codecs 22
VLAN 23
DiffServ 25
Packetization **Error! Bookmark not defined.**
Jitter and Jitter Buffer 25
Firewall and NAT Traversal 26

Glossary of Terms and Abbreviations **28**

Contacting Malden Electronics Ltd **29**

End User License Agreement

In the End-User Agreement below, the following terms are employed with these definitions:

- The End-User is the person or entity that has purchased the Software and is using it in the course of its business.
- An Affiliate is a legal entity directly or indirectly controlling, controlled by or under common control with an End-User. Control of an entity shall exist through the direct or indirect:

  a) control of 50% or more of the nominal value of the issued share capital of the entity or of 50% or more of the entity's shares entitling the holder to vote for the election of directors or persons performing similar functions,

  or

  b) right by any other means to elect or appoint directors of the entity (or persons performing similar functions) who have a majority vote.
- The Software is the application program and related programs including the speech quality metric(s) and firmware that have been supplied to the End-User and have been installed from the original media along with any product enhancements supplied by Malden Electronics Limited and come into being upon a single computer platform that is connected to a single Digital Speech Level Analyser (DSLA).

By installing and using the Software the End-User agrees to the following:

1. The Software and firmware shall only be used in conjunction with the DSLA.
2. One instance of the Software may be caused to come into being on one computer platform.
3. The Artificial Speech Test Stimulus (ASTS) may only be used for the purpose of making a measurement or a sequence of measurements, as part of which at least one such copy of the ASTS shall be processed through the Software. No more than five simultaneous copies of the ASTS may be being generated at any one time.
4. The End-User shall not copy the Software in whole or in part, other than is essential for the proper operation of the Software or for normal security back-up purposes.
5. The End-User shall not modify, translate, reverse-engineer or decompile the Software except to the extent permitted by law.
6. The End-User shall maintain the Software in confidence and ensure that it is protected from unauthorised copying or disclosure by measures that are no less stringent than those it uses to protect its own valuable information and that are, in any case, no less than reasonable in the circumstances.
7. The End-User shall prohibit the use of the Software by anyone other than the End-User, its employees and agents.
8. The acknowledgement of the rights in the Software shall not be removed from the Software or any installation of it
9. The End-User shall not transfer or assign the End-User Agreement except to an Affiliate of the End-User.
10. The validity construction and performance of this Agreement shall be governed by and interpreted in accordance with the laws of England.
11. Acknowledgments - H.323 software uses the Open H.323 stack version 1.14.0. SIP software uses libosip2-2.0.6.

Introduction

What is a DSLA Service?

A DSLA service adds an alternative test interface or external device control capability to the DSLA application.

With an “interface” service the existing DSLA channel is replaced totally by the selected service, while for a “control” service the DSLA channel is supplemented with control messages to an external device (e.g. to make a telephone automatically go on hook when required).

This manual describes the interface services which are collectively called *Network Voice Services*.

What is NVS?

Network Voice Service (NVS) is an installable option for DSLA users to allow direct connections to VoIP and ISDN networks. With NVS end-to-end testing using analogue interfaces can now be extended to include digital origination and termination points.

NVS allows network staff to place a "virtual phone" anywhere inside the network, enabling measurements to be made between a physical end point and a (movable) virtual point.

Using the DSLA Scheduler, each channel in the DSLA user interface can be configured to control one of the following services:

- H323
- SIP
- ISDN

To make use of the NVS, you must first install the application on either the DSLA application PC or a networked Windows PC.

Multiple NVS Installations

There are two automatically detected modes to support the use of multiple NVS installation. These modes are:

Consecutive Mode

Consecutive mode allows a licensed NVS to be installed on multiple PCs around a network but only a single instance of NVS to be used by a schedule. When you connect to the licensed NVS on another PC you will be prompted by the DSLA User Interface to confirm the change.

Concurrent Mode

Concurrent mode allows the use of two or more instances of NVS in a single schedule. Each instance is installed on a different PC using a unique installation CD.

Getting Started

The following section hopes to provide enough information to get a NVS installation making and assessing phone calls across a network. This section covers:

- System Requirements
- Installation
- NVS User Interface
- Quick Start Example
- Using your Playlists and Schedules

System Requirements

Before you install the software, you should make sure that your computer meets the minimum system requirements.

- Windows 2000/XP
- Pentium-class PC (1 GHz or higher recommended)
- 128 MB RAM

Installation

1. Insert the CD in the CD-ROM drive. The installation process should start automatically. However, if it does not, then follow these steps:
   - From the Windows Start Bar, click Run.
   - Type <DRIVE>:\Setup.exe where <DRIVE> is the CD-ROM drive you are using and click OK.
2. The setup program prompts you through the installation process. Follow the instructions on the screen.

NVS User Interface

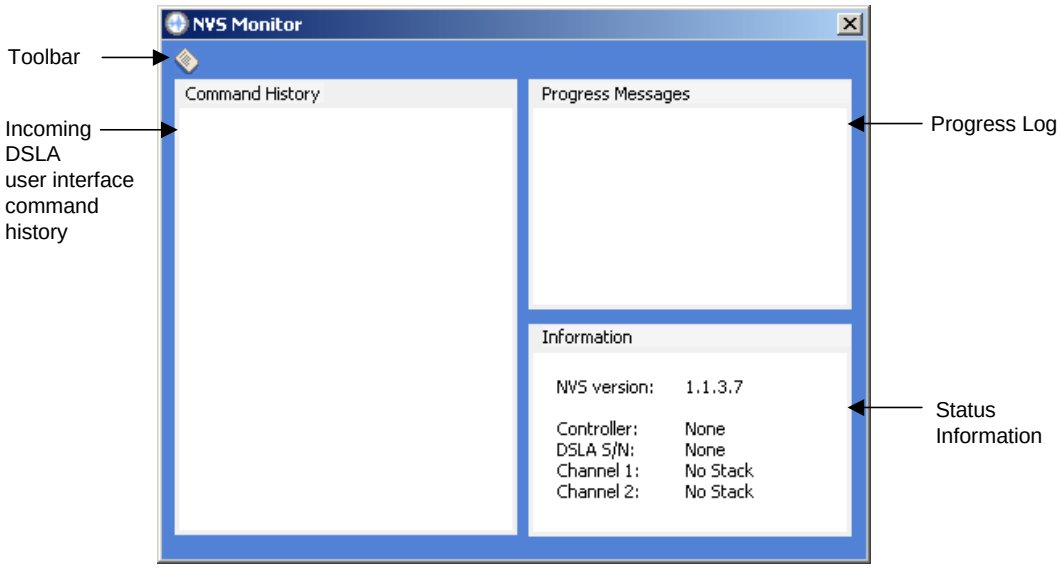
NVS is controlled from the DSLA User Interface. The NVS Monitor window allows the user to monitor progress and set logging levels.

The NVS software starts up with Windows and runs as a background process. When the software is running, you will see the NVS icon in the system notification area.

Right-clicking the NVS icon will display a menu that will allow you to:

- Open the NVS monitor window
- Toggle logging on and off
- View the log file
- View the Malden Electronics web site
- Exit the NVS application

Double-clicking with the left mouse button on the icon will also open the NVS monitor window as show below:



A single toolbar icon is shown at start, clicking on this icon will start NVS logging and extend the toolbar to show all log buttons.

Note: Closing the monitor window will only minimise the NVS Monitor window to the system notification area. To close the NVS application, right-click the NVS icon and select **Exit** from the menu.

Quick-Start

The easiest way to start using an NVS installation is to make use of a Quick-Start example. The DSLA User Interface offers a number of Quick-Start schedules to help users familiarise themselves with DSLA testing. These Quick Start schedules are located under the "Help | Quick Start Examples" menu, and when selected load an example schedule into the DSLA.

For a lot of users, the Quick-Start schedules require only a little editing to achieve first time results. The latest version of DSLA User Interface is shipped with 6 Quick-Start schedules designed to demonstrate NVS functionality:

Quick Start	Description
H323 to NetMeeting	NVS calls a NetMeeting client and if a call is established plays speech and then terminates the call
H323 to H323	call between two NVS H323 instances and assesses the connection quality, terminating the call at the end
SIP to Messenger	NVS calls a Microsoft Messenger SIP client and if a call is established plays speech and then terminates the call
SIP to SIP	call between two NVS SIP instances and assesses the connection quality, terminating the call at the end
ISDN to phone	NVS calls a phone number and if a call is established plays speech and then terminates the call
ISDN to ISDN	call between two NVS ISDN instances and assess the connection quality, terminating the call at the end

5 Steps to Quick Start

1. Open the DSLA user application and ensure you are connected to the DSLA for which the NVS instances are licensed.
2. Select the desired Quick-Start example from the "Help | Quick Start Examples | <Language>" menu. <Language> is either "American" or "British"
3. Locate the DLSA Scheduler (Tools | Scheduler) and click on the first "A calls …" task. This will cause the User Interface to try and connect to an NVS instance. The schedule holds the IP address for each NVS configured in a schedule. The DSLA User Interface will try to connect to this address for up to 10 seconds, if it is unable to find the NVS in this time a dialog box will be raised. In the dialog box you can directly enter the IP address of the required NVS instance or if an NVS is on the same subnet you can browse for an NVS instance. (This may occur for Channel A and Channel B)
4. Locate the Channel A window (View | Channel A) and double-click on the event in the Playlist which shows the telephone icon; historically this played DTMF digits but is now also used to hold the phone number or IP address or username for NVS calls.

An “Event Properties” dialog box will appear, edit the telephone number and click Apply and then Close. Save the Playlist, overwriting the previous Playlist and save the Schedule.

5. Now click the “start schedule” button in the toolbar of the DSLA scheduler.

[1]Number format guidelines can be found in the How-To section.

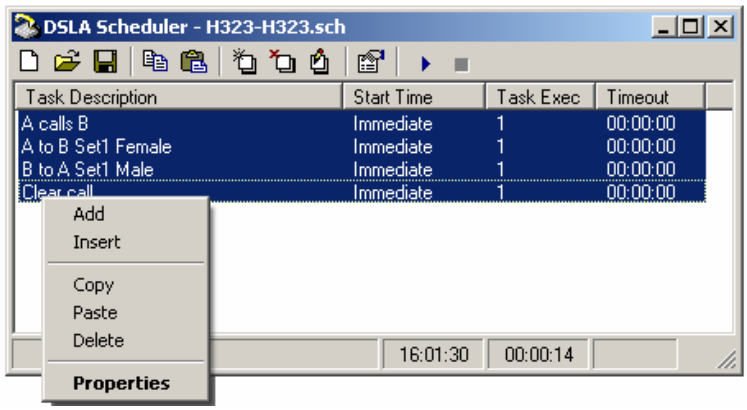
Using your Playlists and Schedules

NVS works with most playlists although not all playlist-events are supported (see the end of this section for more information). To use NVS a schedule needs to be configured to set either Channel A or Channel B or both to a service.

Configuring Schedules

A channel is set to use a Network Voice Service by configuring the Channel Settings for each task within a schedule. To configure a schedule to use a NVS follow the steps below:

1. In the DSLA scheduler select the tasks which you wish to configure, click the right mouse button and select “Properties” from the pop-up menu.

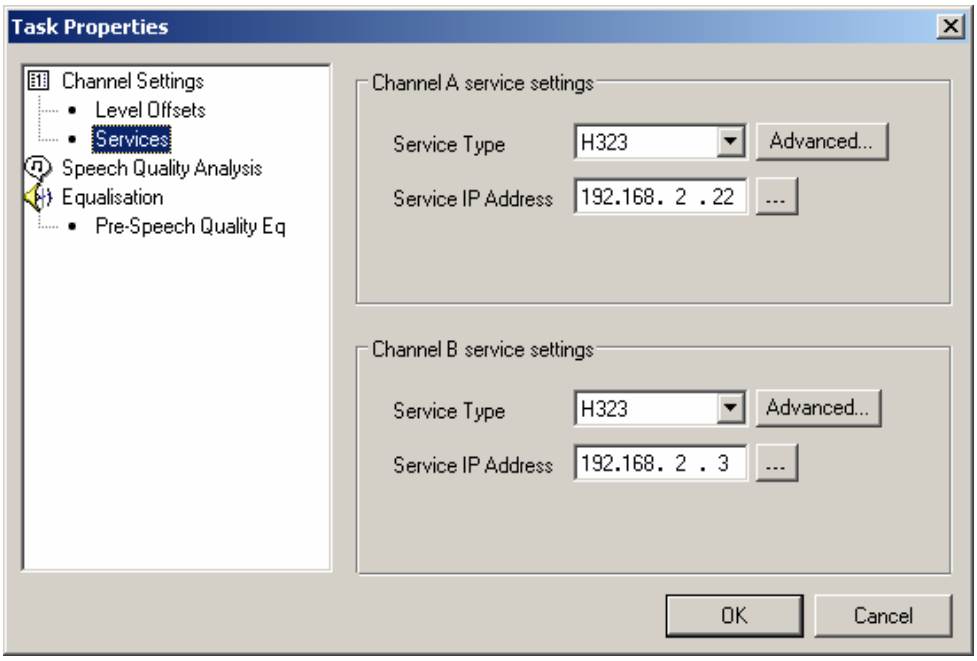


Tip 1: Select all tasks by pressing Ctrl-A

Tip 2: Select multiple tasks by selecting a first task , holding down the Shift key, and selecting the last task.

Tip 3: Stop the User Interface from trying to connect to a service by selecting a task with the right mouse button rather than the left.

2. After selecting “Properties” in the pop-up menu the **Task Properties** dialog appears. Select **Services** in the tree view on the left hand-side of the dialog in order to be able to configure the service type for both Channel A and Channel B.



Loading Playlists

To use a playlist with Network Voice Service it must be saved and loaded using an NVS configured schedule. If no schedule has been loaded the playlist will use the DSLA analogue ports.

Playlist Event Limitations

Commands that are not support are ignored; this can lead to timing issues, if for example, the playlist does not progress until a particular signal has been received.

The currently supported playlist events are:

Playlist Event	Comments
Speech Events	Supported
DTMF Events	Places call rather than plays DTMF
Control Events:	Limited to those listed below:
Start Measurement	
Stop Measurement	
Phone On Hook	
Phone Off Hook	
Wait	
Wait – Ring Detect	No actual ringing, but answers incoming call
Speech Quality Analysis	
Silence	
Execute Application	

How To

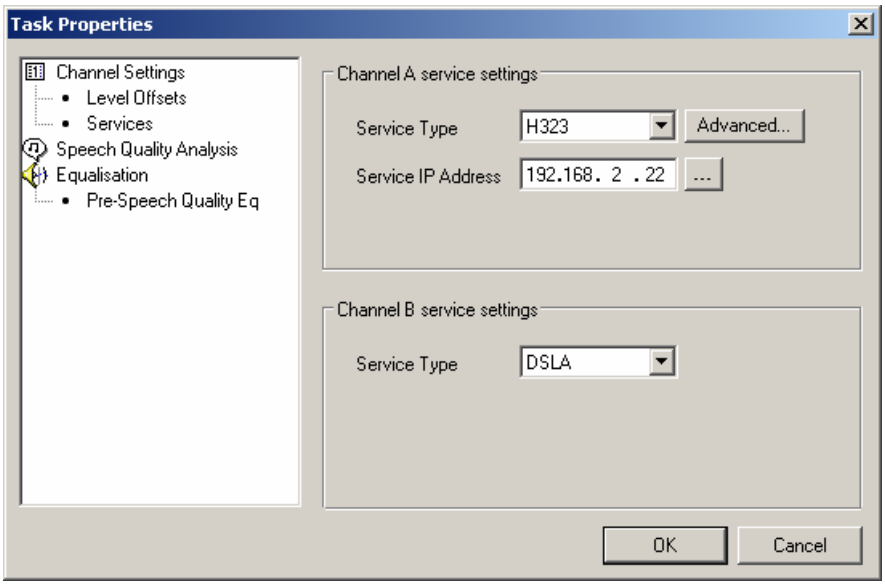
How to make a H323 (NetMeeting) call

Assumptions

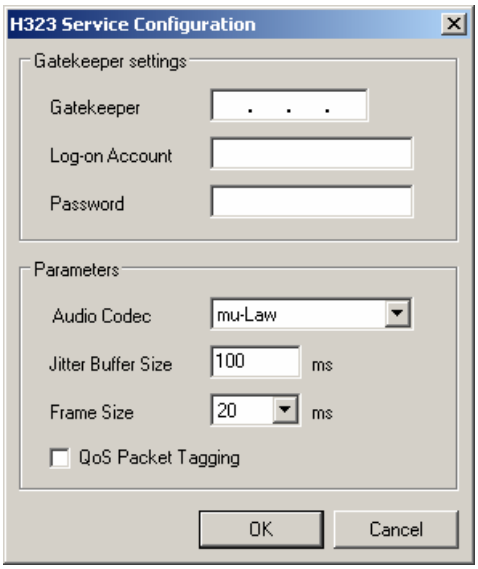
It is assumed that NetMeeting and NVS applications run on different PCs. H323 requires the applications to use port 1720 and two applications on the same PC cannot share this port. If both applications are running one application will be able to use port 1720 while the other will not.

Service Settings

The H323 service type is selected and the IP address of the NVS is set for each task in the schedule.



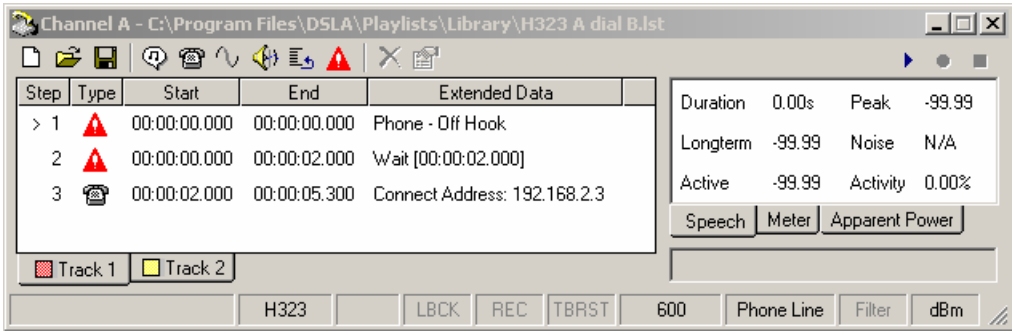
The defaults for codec, jitter buffer and frame size can be adjusted by clicking on the **Advanced…** button. The gatekeeper settings should be left blank as shown below:



The QoS Packet Tagging option is described more fully in a later section.

Phone Number Format

To call NetMeeting or another H323 endpoint you enter the address of the destination host (a machine name such as “h323.malden.co.uk” or a dotted number such as “192.168.2.3”) in the DTMF event.



How to make Cisco Call Manager routed calls

Assumptions

Cisco IP phones setup calls via a Call Manager using a propriety protocol called Skinny. To call these phones NVS needs to place an H323 call to the same Call Manager. This requires the NVS machine to be registered with the Call Manager as an H323 phone or gateway.

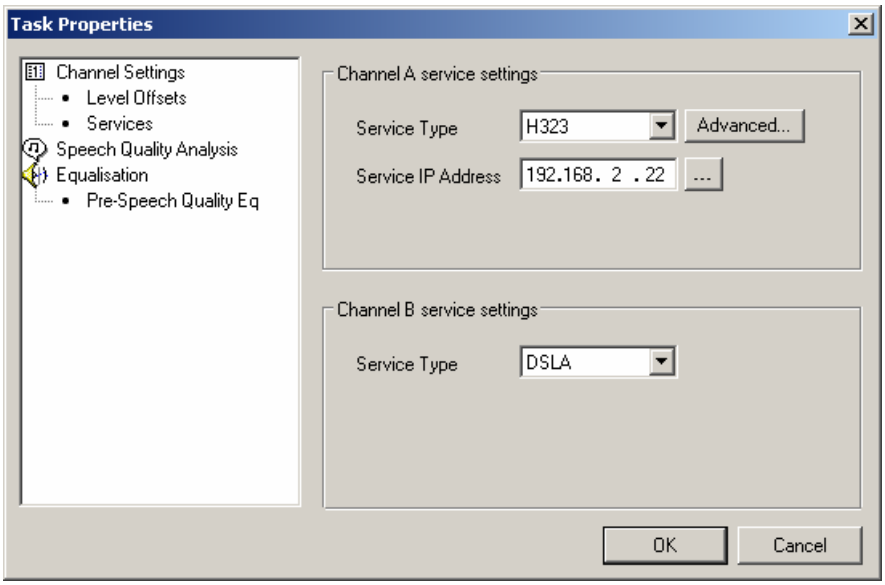
Call Manager Configuration

To register NVS with the Call Manager as an H323 phone follow the steps below:

1. In the **Call Manager** web interface, Select **Device | Add a New Device** from the main menu.
2. Select **Phone** from the drop-down menu and click **Next.**
3. Select **H.323 Client** from the **Phone type** drop down list and click **Next.**
4. Fill in the **Phone Configuration**, a minimum configuration example is described below:
5. **Device Name** needs to be the *IP address* of the NVS machine.
6. **Device Pool** select the same pool as other IP phones that you will test.
7. To assign NVS with a phone number enter this in **Outgoing Caller ID Pattern.**
8. When configured click **Insert.**

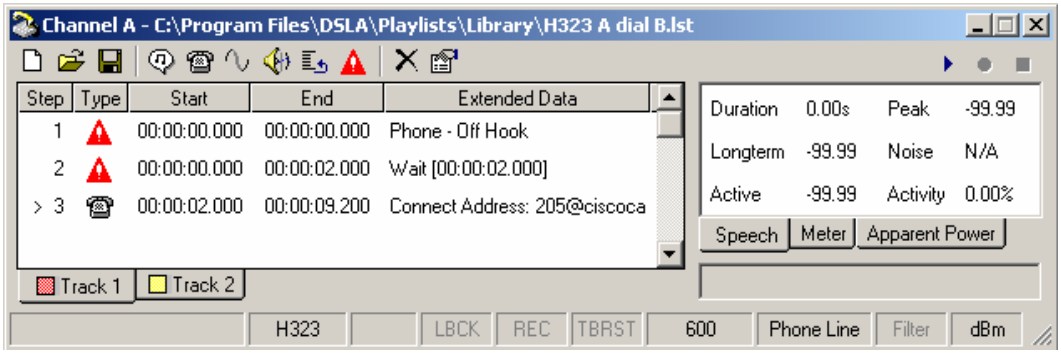
Service Settings

The service is set up in the same way as the previous H323 NetMeeting example. The H323 service type is selected and the address of the NVS is entered or browsed.



Phone Number Format

To place a call via a Call Manager requires the destination number to be pre-pended to the address of the call manager in the format “destination@callmanager”(a call manager address can be in the form “callmanager.net” or a dotted number such as “192.168.2.16”)



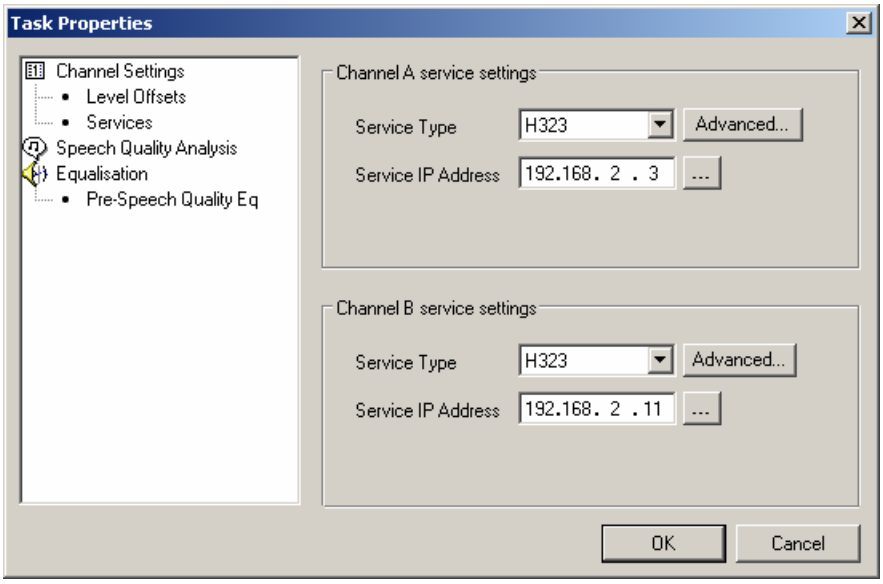
How to make H323 gatekeeper routed calls

Assumptions

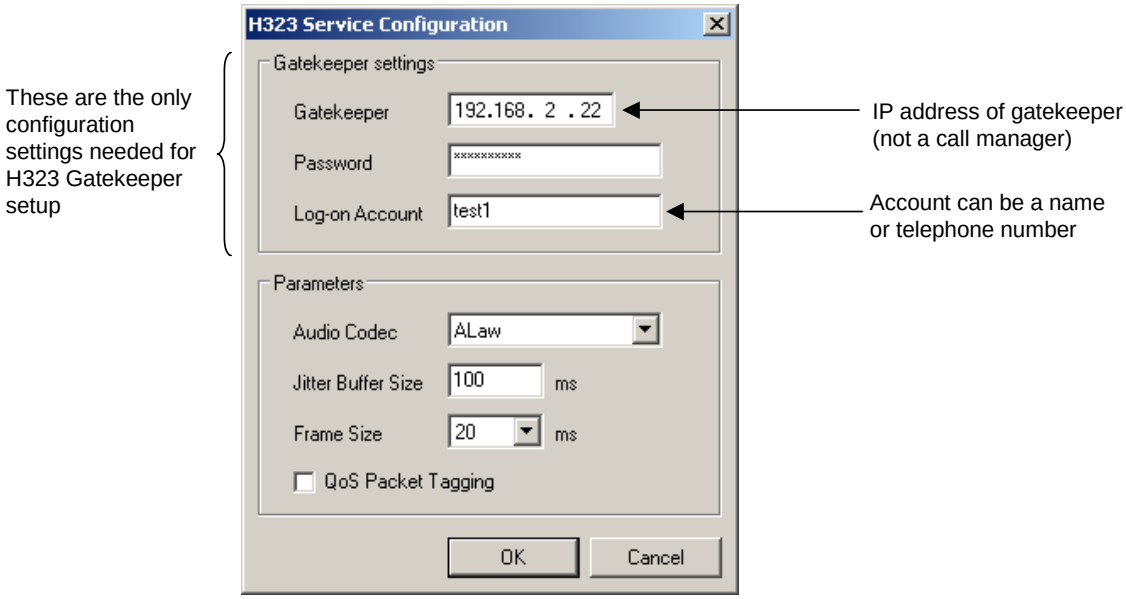
For this example it is assumed that you want to make a NVS to NVS gatekeeper routed call. If the destination is not to a NVS then you need only configure the channel that makes the call.

Service Settings

Gatekeeper settings are configured for a task under the Task Properties “Advanced…” button:

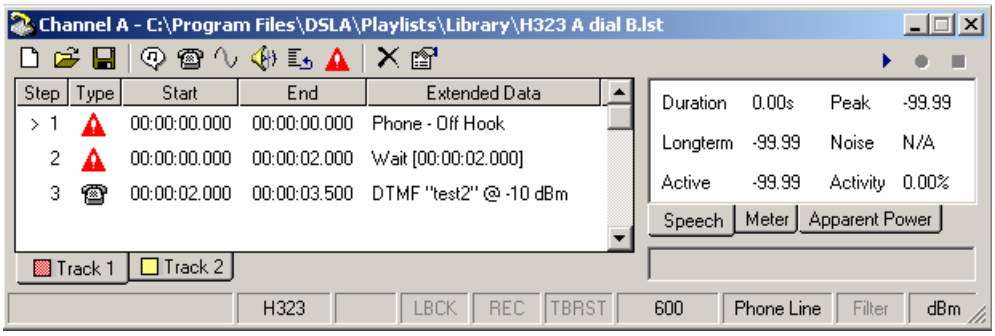


Clicking on the **Advanced…** button opens the dialog below:



Phone Number Format

If you are gatekeeper registered and you wish to make a call it is expected that the call will be routed via the gatekeeper. As such you should provide the destination identifier that the gatekeeper will use to route the call as the dial string, for example the destinations telephone number or a user name.



Note 1: Replace ‘test2’ with a phone number or H323 username

Note 2: You are unlikely to specify an IP address as a dial string for Gatekeeper routed calls

How to call MS Messenger (SIP)

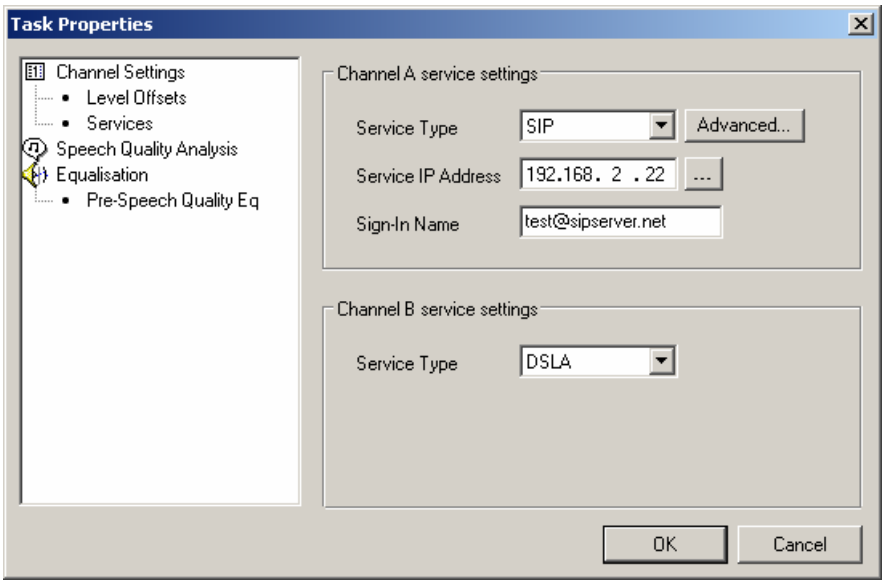
Assumptions

It is assumed that the version of your Microsoft Messenger supports SIP based phone calls and that its **SIP Communications Service Account** settings have been configured in the **Tools | Options | Accounts** panel.

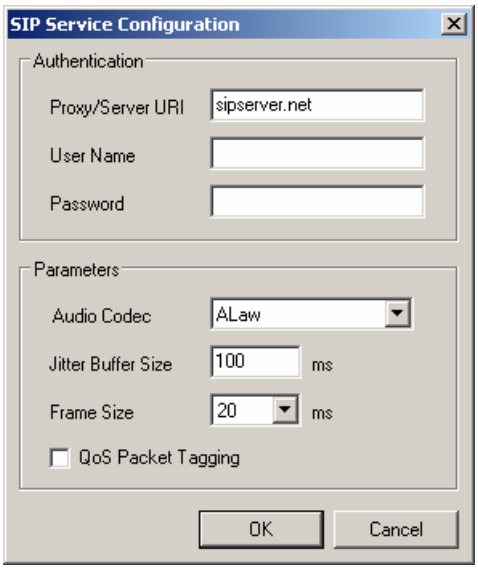
Service Settings

NVS is required to register with a SIP server or proxy in order to be able to place a call. The server address can be configured in two ways:

The address of the SIP server is placed as the domain of the **Sign-In Name** (user@domain)

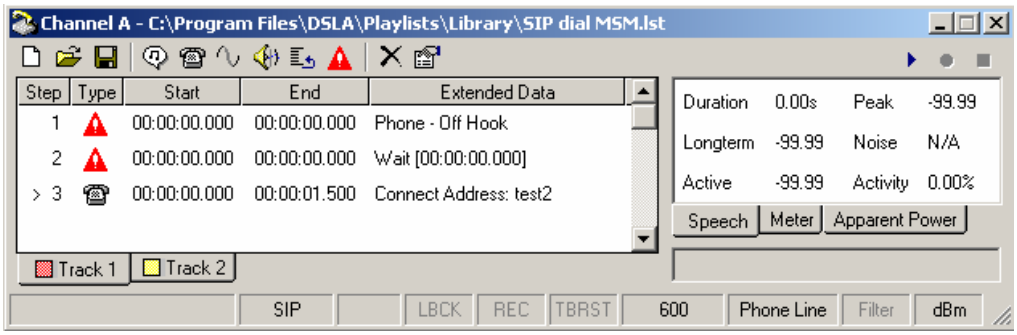


or if this is not possible (e.g. the Sign-In Name is a user's email address) the address is entered under the **Advanced…** dialog box



Phone Number Format

The destination identifier for SIP can be anything from a phone number, to a name to an email address. All valid combinations are supported and should be entered via the DTMF playlist event.



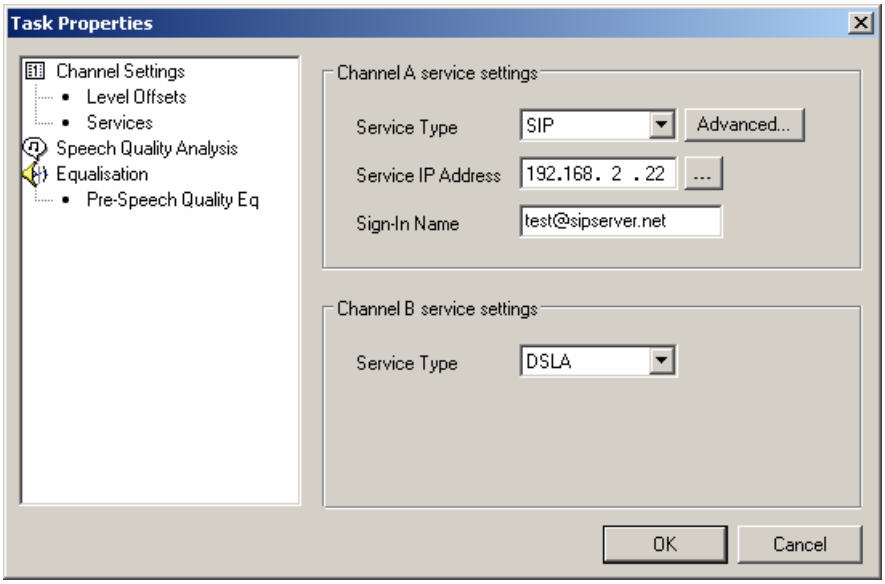
How to make SIP calls with authentication

Assumptions

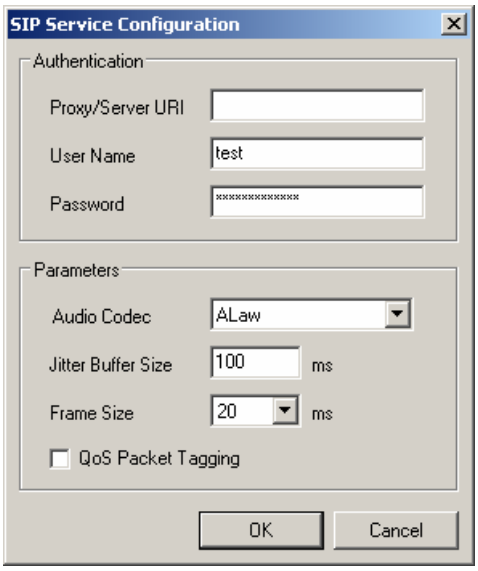
It is assumed that the **Sign-In Name** contains the address of the SIP server as the domain name. If this is not the case refer to the previous example to understand how to set a different address for the SIP server that that of the Sign-In Name domain name.

Service Settings

The service to be set up in the same way as the previous SIP MS Messenger example. The SIP service type is selected and the address of the NVS is entered or browsed.

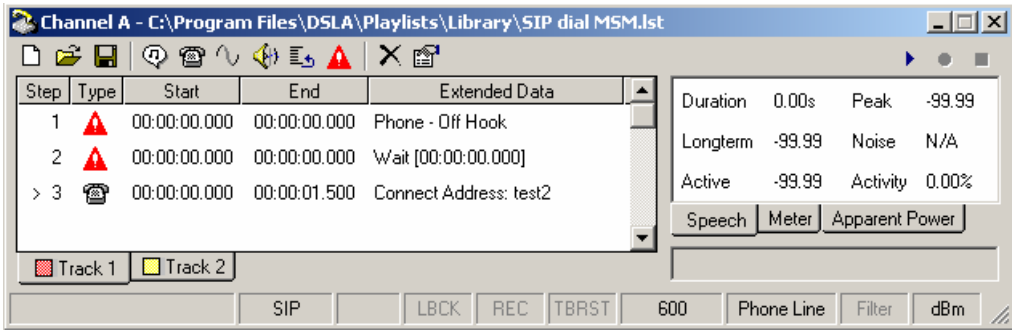


In the **Advanced…** dialog box the Authentication **User Name** and **Password** are entered. Note the authentication "realm" is not entered as it will be extracted from the server authentication request.



Phone Number Format

The destination identifier for SIP can be anything from a phone number, to a name to an email address. All valid combinations are supported and should be entered via the DTMF playlist event. This is equivalent to how the previous example's destination is configured.



Frequently Asked Questions

Connecting Issues

Q. Why do I get “not licensed” message?

A. If you have the DSLA connected via the serial port and not via the Ethernet port to the DLSA user interface PC the NVS will not work.

or

The DSLA user interface shows available NVS’s when browsing from the scheduler task property, it will allow you to select an NVS, but will then fail its licence check and report, “You are not a licensed user of the selected service. Please contact Malden Electronics Ltd to obtain a licensed installation.”

or

The connect between the DSLA and the DSLA user interface is unreliable. Try replacing the network cable.

Q. When I open the NVS browse window I cannot find my installed NVS

A. Only NVS’s running on a machine in the same subnet will appear in the browser. If you do not see the machine you are trying to connect to, it may be that the machine is in a different subnet. Try entering the IP address of the NVS machine directly into the “Service IP Address” field in the Scheduler Task Property and then selecting the task in the scheduler to see if it connects.

Codecs

Q. What codecs are supported by the VVT?

A. At this time, the VVT supports G.711 Alaw and μlaw.

Q. What are the most commonly deployed VoIP codecs?

A. The most popular and commonly deployed VoIP codecs are G.711 and G.729. G.711 is used primarily within campus and LAN installations with G.729 being used for WAN and Centrex installations.

VLAN

Q. Does the VVT support VLANs?

A. Yes, VVT supports the main standard of 802.1Q/P

Q. Why are VLANs important to VoIP?

A. VLANs are being used to distinguish VoIP traffic from data on a network and allow traffic shaping to ensure that VoIP gets priority. Many IP Phones are marking voice streams with an 802.1P bit setting of 5. It should be noted that older network switches and routers may drop these frames.

Q. What does 802.1Q/P signify?

A. 802.1Q specifies the addition of a VLAN tag that is inserted into the Ethernet frame. According to 802.1Q, the tag has two parts: the VLAN ID (12 bits) and Prioritization (3 bits). Because the Prioritization field was never defined in the 802.1Q standard, the 802.1P was introduced to define Prioritization levels.

Q. What do we need to do to enable VLAN support?

A. This only works on Windows 2000 and XP and is implemented as part of the QoS Packet Scheduler network service. The basic procedure to add VLAN support is the same for Windows XP and 2000, although 2000 has a few additional steps:

Windows XP and 2000

1. Click Start > Settings > Network Connections
2. Double-click Local Area Connection and click Properties
3. On the General tab click the Configure button.
4. On the “Advanced” tab select the “QoS Packet Tagging” property and make sure the value reads enabled.

Windows 2000 additional steps

1. Add the Packet Scheduler service by, Click Start > Settings > Network Connections
2. Double-click Local Area Connection and click Properties
3. Click Install. Highlight Service and click Add
4. Click Microsoft and select install QoS Packet Scheduler

Q. I can’t find “QoS Packet Tagging” option, where is it?

A. This may be for one of three reasons:

- “QoS Packet Tagging” is not a Windows defined name and different network card manufacturers may use different names, such as “802.1p QOS”, to describe the option.
- The device driver may need updating, try installing the latest driver from the network card vendor.
- The capability may not be supported on your network card; unfortunately even with new network cards this feature is not a Windows requirement. This feature tends to be left off low-cost NICs.

Q. I get RSVP messages, how can I stop these?

A. Windows 2000 implements RSVP bandwidth reservation protocol as part of its QoS Packet Scheduler service, for most VoIP testing this should be disabled. In Windows XP this is disabled by default.

To disable RSVP in Windows 2000 requires two registry keys must be added/changed:

`HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\QoSSP`

Entry name	Data type	Value
`EnableRSVP`	`REG_DWORD`	0
`EnablePriorityBoost`	`REG_DWORD`	1

Q. Can we set the VLAN priority bits to something other than 5?

A. Yes, but this requires another registry edit.

`HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\Psched\Parameters\Adapters\<Adapter-GUID>\`

Entry name	Data type	Value
`ServiceTypeGuaranteed`	`REG_DWORD`	0 – 7

Q. Can we set the VLAN ID to something other than its default?

A. With some network cards this is set in the same place that QoS Packet Tagging is turned on/off. The automation of this is under investigation.

Q. With QoS on why can I only hear speech in one direction, even when I can see the packets using Ethereal?

A. If the test works without QoS enabled and you can see the packets arriving at the termination point with a packet capture tool such as Ethereal, the most likely reason you can't hear anything is that the NIC on the terminating machine does not support 802.1Q. Try upgrading your NIC drivers or using a different machine / card.

Q. My NIC supports 802.1Q but I can't see the tags when I capture the packets, why?

A. Windows 2000 network driver architecture contains four types of kernel-mode network drivers.

- Miniport – directly manages a NIC and provides interface to higher level drivers
- Intermediate – interface between upper-level protocol such as a legacy transport driver and the miniport driver
- Protocol – for implementing a specific protocol
- Filter-hook – can be used to filter packets

With your NIC it is likely that the tagging will be implemented in the miniport driver or potentially in a intermediate driver. The Windows specification for both types of implementation state that upon receiving an 802.1Q marked packet the driver should remove the priority and VLAN ID values from such packets. This information is place in the 'per-packet information' structure before indicating that the packet has arrived.

DiffServ

Q. Does the VVT support DiffServ?

A. Yes, selecting the QoS option in the service property turns this on.

Q. What is DiffServ?

A. A relatively new IETF standard used to indicate a packet's priority level using the IPv4 packet header's Type of Service (TOS) fields. Although many application and router vendors previously used the TOS field to indicate the service level, no standard existed for interpreting TOS tags. The DiffServ standard (RFC 2472) renames the 8-bit TOS field the DS (DiffServ) byte and restructures that field.

Q. What's the difference between DiffServ and 802.1Q?

A. DiffServ information is stored in the IP packet header. The 802.1Q packet tagging is held in the Ethernet frame. DiffServ can be used to carry QoS information between LANs while 802.1Q tagging is very much used to carry QoS information between network interface cards.

Q. I see DiffServ marking but not VLAN tagging, why?

A. The most likely reason is that you have not configured your network card to use 802.1Q tagging, please refer to previous section

Q. Why does my DiffServ marking not arrive at the destination?

A. One possibility is that a network element such as a router is reclassifying your packets and changing the DiffServ marking from one setting to another.

Packetisation

Q. What size packet is used?

A. This is user configurable at sizes of 10, 20, 30, 40 and 50ms.

Jitter and Jitter Buffer

Q. What type of jitter buffer is implemented in the VVT?

A. The VVT uses a static jitter buffer model where the size is user configurable and can be changed during a test sequence.

Q. Why do I see jitter leaving the VVT?

A. The time at which packets are played out onto the network relies on MS Windows timing which can become interrupted by other processes. This is compensated for over the length of a phone conversation to create an average jitter of 0ms but the time interval between two packets may occasionally show localised jitter.

Firewall and NAT Traversal

Q. Will VVT work with Firewalls?

A. Voice over IP (VoIP) will not pass through most of the firewalls installed today without network managers opening up ports in the firewall. While solutions to the problem are emerging, they are not widespread.

Q. Will VVT work with Network Address Translation (NAT)?

A. Like firewalls, the vast majority of installed NAT devices will stop VoIP from connecting media streams even if the call signalling appears to work. Solutions are being developed and deployed, but these are not widespread.

Q. Why do firewalls stop VoIP?

A. Firewalls protect a network by blocking incoming traffic based on application port numbers. A standard configuration closes all ports except those specifically needed (for example port 80 to allow HTTP).

If you want to let VoIP traffic move from one IP network to other through a firewall, you have to leave a range of ports open. To establish a voice conversation with H323, TCP port 1720 must be opened, while for SIP UDP port 5060 must be open. Then, for the voice traffic itself, a UDP port must be opened for the RTP voice traffic and, optionally, a UDP port may be opened for RTCP.

RTP ports are chosen at random and can be any number above 1024, with the RTP on an even port number and the corresponding RTCP port one above, e.g. 5010 for RTP and 5011 for RTCP control information. For each call that you wish to have running, you'll have to open a set of ports.

The solution is to have firewalls open VoIP ports dynamically when VoIP traffic needs to get through, and closed them once the call is finished. This requires the firewall to understand enough of the VoIP call control to know which ports each call will use. Today, several firewall vendors, including Cisco and Check Point have introduced products that allow dynamic port opening for VoIP.

Q. Why does NAT stop VoIP?

A. NAT allows a single device, such as a router, to act as an agent between one network (for example public Internet) and a local (or private) network. This means that only a single, unique IP address is required to represent an entire group of computers. For most Internet communications this is fine, when a packet traverses a NAT element the source and destination IP addresses in the headers are mapped to the correct representation. However, for VoIP packets there is also a need to map some IP addresses held in the payload (the call signalling). If this mapping is not applied it becomes impossible for the two ends to achieve a full end-to-end connection of the media paths.

Q. Who develops Firewalls and NATs that allow VoIP?

A. Companies claiming to have firewall and NAT products that support H323 and SIP based VoIP include:

- Cisco
- Check Point
- Netrake
- Edgewater Networks

Glossary of Terms and Abbreviations

Codec	*Co*der *Dec*oder for voice stream compression, e.g. G.711
ITU	International Telecommunications Union
IEEE	Institution of Electrical and Electronic Engineers
RTP	Real Time Protocol
VoIP	Voice over Internet Protocol
URI	Uniform Resource Identifier - the W3C's codification of the name and address syntax of present and future objects on the Internet. In its most basic form, a URI consists of a scheme name (such as file, http, ftp, news, mailto, gopher) followed by a colon, followed by a path whose nature is determined by the scheme that precedes it (see RFC 1630). URI is the umbrella term for URNs, URLs, and all other Uniform Resource Identifiers.
SIP	Session Initiation Protocol
SIP Proxy	A SIP proxy server receives SIP messages from user agents and acts on their behalf in forwarding or responding to those messages. Their function is not unlike a gatekeeper or call agent in other voice-over-IP environments. Also, proxy servers add services, features and scalability to SIP networks. The SIP proxy server typically includes a registration service and a SIP location database, in addition to the SIP proxy.

Contacting
Malden Electronics Ltd

Malden Electronics Ltd
2 High Street
Ewell
Surrey
KT17 1SJ, England

Tel: 020 8786 9145 International: + 44 20 8786 9145
Fax: 020 8393 6883 International: + 44 20 8393 6883

Web: www.malden.co.uk

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