Miercom

Lab Testing Summary Report

March 2012

Report SR120220B

AR Series Routers:

Voice Features and Performance

Vendor Tested:

HUAWEI

Products Tested:

AR207V-P
AR1220VW
AR3260
Enterprise Routers

Key findings and conclusions:

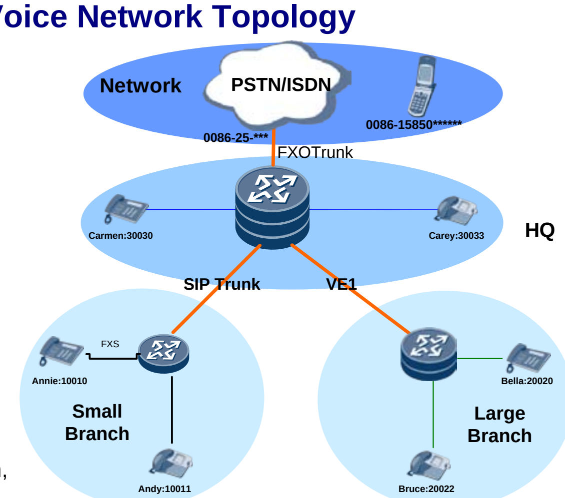
- **Huawei AR G3 routers support SIP, PRA, and ATO voice trunks**
- **Voice services supported include 3-way conference calling and simultaneous/sequential ringing**
- **During routers SIP call completion tests AR G 3 routers achieved 100% call completion with no dropped calls**
- **AR G3 routers have built-in PBX voice functions, including conference calling and multi-language IVR**

Huawei Technologies engaged Miercom to evaluate several series of enterprise routers for voice capabilities and features. The features are common to the AR 200, AR1200 and AR3200 Huawei router series product line. The features and capabilities discussed in this report by product are supported in the entire family of products.

The AR200 series of routers has a rich set of voice functions, including basic calling, voice trunking, and 3-way calling. These routers are for use in branch offices or small business deployments. The AR200 series are fixed port devices without service slots for interface cards.

Plug-and-play configuration and deployment is supported by using a USB flash drive. Building an enterprise network can be expedited, and IP addressing does not need to be configured manually. The AR200 series is a fixed configuration, offering eight FE ports, and two WAN uplinks, which provide both load balancing and link redundancy. The AR200 series also reduces office noise with its fan-free design, and can be deployed in a severe environment, thanks to a 6kV surge protection capability.

Figure 1: Huawei Enterprise Routers Voice Network Topology



Source: Miercom, March 2012

Three different office locations were simulated for voice testing, connected to a headquarters location. Voice quality, features assessment and overall QoE were rated exceptional.

The AR1200 series of routers has four models: the AR1220, 1220V, 1220W, and 1220VW. The 1220V model has voice features and DSP with 32 supported channels. The 1220W is a wireless router with 802.11 b/g/n capabilities.

The AR1200 series of routers has two card slots for interface cards supporting different network uses and deployments. These cards include Layer 2 and Layer 3, FXO voice, and E1/T1 multifunction cards.

The AR1220W and 1220VW support wireless capabilities for Wi-Fi routing and 3G networks. The AR1220V and 1220VW have voice capabilities with a built-in IP PBX, making it ready to use for UC without additional hardware.

The AR1200 series is highly reliable and stable with hot-swappable interface cards and redundant fan modules.

The AR3200 series of routers provides routing, switching, voice, security, and wireless capabilities such as 3G and Wi-Fi. These routers are used in an enterprise environment such as headquarter offices. The AR3260 router has six card slots for Layer 2 / Layer 3, FX0 voice, and serial port cards. The AR3260 can also support up to 147 GE ports using the six available card slots. There are 24 ports on each of the removable cards and three additional static ports on the front of the router itself. The AR3200 series can also support multiple tunneling protocols such as VPN and GRE.

Voice features for the AR3200 include a built-in IP PBX. As a result of having an embedded PBX, voice features can be utilized without the need for additional devices or hardware.

The AR3260 router supports a large number of SIP users on its PBX system. The AR3260 can support up to 2,048 users at a time, which is twice the number of users the AR1200 series of routers is capable of supporting.

Basic Voice Calling

This type of call testing was designed to test the basic call functionality of the AR G3 routers. Basic calls needed to be verified from both FXO lines and IP phone lines.

Using the same topology diagram as in *Figure 1* on page 1, voice calls were placed with phones connected via FXO lines and between IP phones using the embedded PBX within the router.

During the testing, calls were placed between the phones and PBX features were verified. Basic phone features support were verified including call busy, call hold, call park and call forwarding. These calls were able to be answered and disconnected without any problems detected. This verifies the basic call functionality of the AR G3 routers.

Voice Trunking

Designed to exercise the voice trunking capabilities of the AR G3 routers, our tests included verified a variety of different trunks, such as SIP, PRA, and ATO (FX0). To test these trunks, a large multi-location deployment was set up. The topology in *Figure 1* on page 1 shows the set-up that was used for the voice trunking portion of testing.

There was one type of trunk set up between each of the locations, which allowed the features to be tested using one large network deployment. For testing the SIP trunk, calls were placed from the small branch location to the headquarters location. These calls were all placed, answered, and disconnected properly without any problems detected on the system. Since the only connection between the small branch office and the headquarters was a SIP trunk line, this verified that the SIP trunk portion of the routers worked.

To test the PRA (VE1) trunk, calls were placed from the large branch location to the headquarters location. Again, calls were able to be placed, answered, and disconnected without error.

Finally, to test the FXO line, calls were placed from the headquarters location to the PSTN network. Calls were placed, answered, and disconnected without error.

All three types of voice trunks tested with the Huawei AR G3 routers worked properly and as expected. We did not encounter any errors or issues during testing of the Huawei products for this review.

Additional Services

Additional voice features of the AR G3 routers were tested, including three-way conference calling, simultaneous/sequential ringing, and Interactive Voice Response (IVR). These features were all tested using phones connected to a single AR router.

To test the three-way conference call, a mix of PSTN and IP phones was connected to the AR router. A call was placed from an IP phone to a second IP phone. Once the call was answered, it was placed on hold while a third caller was added to the conference call. The third caller in this test was a using a PSTN phone. This test was repeated several times, with each phone taking a turn initiating the conference call to verify that the calling order does not matter. Conference calls could be placed within one networked office as well as between two different office environments. Calls were placed, answered and conferenced easily and voice quality was acceptable with no errors reported.

Simultaneous/sequential ringing allows different notifications based on user requirements. This feature allows simultaneous ringing on multiple phones according to a preset configuration. A sequential ring allows phones to ring in a sequence similar to a failover. When a call is not answered at one station, it rolls over to the next station and the next until it is answered. This is a reliable way to provide consistent call answering.

Interactive Voice Response is now a prerequisite in all PBX systems. The ability to respond to voice interactively eliminates the need for a person to answer calls. The IVR from Huawei has the ability to use multiple languages as well as different voices. The voice selection includes accents from various regions or countries, as well as male or female voices.

Voice Performance

SIP user capacity was evaluated for the AR G3 routers. The AR200 series of routers was able to support a maximum capacity of 16 SIP users. We verified this feature by setting up 16 SIP users and then placing calls using an Abacus 5000. Calls were placed between the various SIP users on the AR207V-P router. Eight of the users placed calls and the other eight users answered calls. This test resulted in 100% call completion rate.

The same SIP user test was performed for the AR1200 series router and the AR3200 series router. Both routers were able to reach their advertised maximum SIP user capacity and both routers achieved a 100% call completion rate. The AR1220 router reached 64 SIP users. 2,048 SIP users were reached by the AR3260 router.

To test the call completion rate of the routers under heavy load, a traffic load generator was hooked up to each AR router respectively and set to deliver a load of traffic to the router while calls were being placed by the Abacus 5000. The traffic generated from the load generator was 128-byte frames being transmitted at 50 Mbps. In all cases, the call completion rate of the routers was 100%.

PoE

We tested PoE support for the AR200 and AR1200 series routers by connecting PoE phones to the AR routers. Both Huawei and Avaya IP phones were used to verify functionality of PoE support. All types of phones worked properly with the Huawei AR routers.

Bottom Line

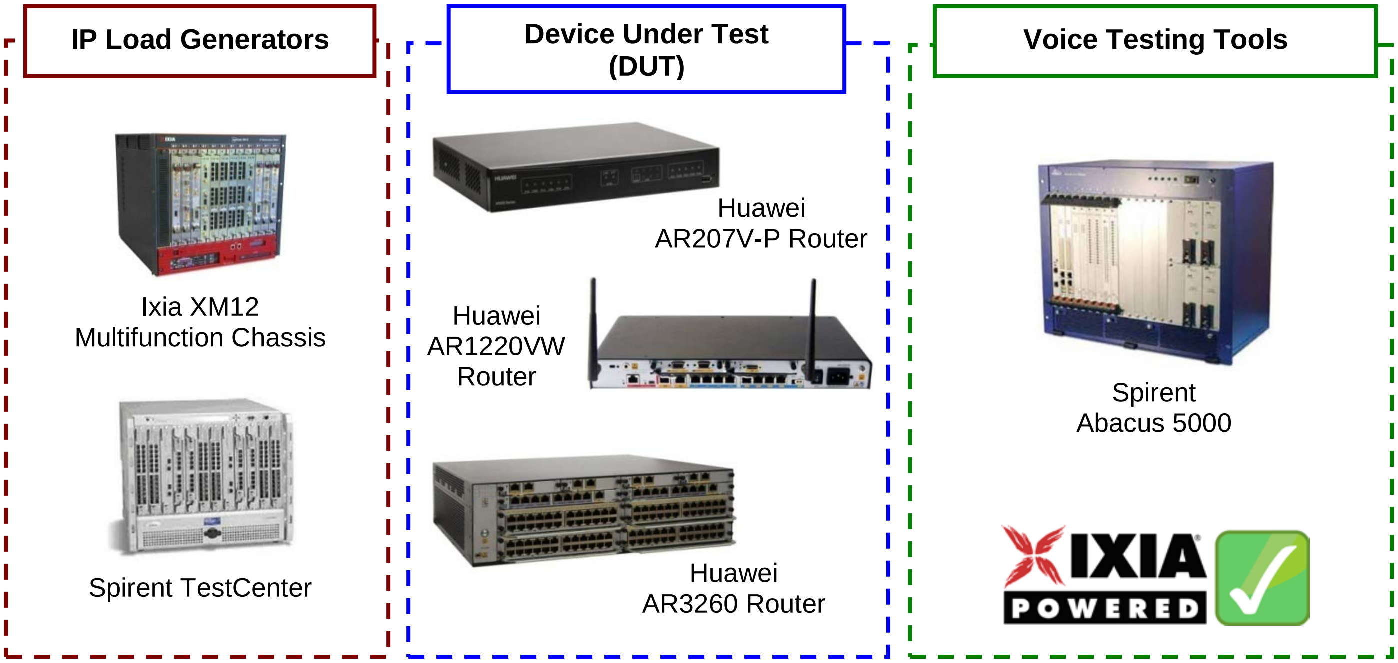
The AR G3 routers with built in PBX provide a rich set of voice features. The AR3260 can support up to 2,048 SIP users at a time, while the AR1220 can handle 64 SIP users.

Voice quality was rated as excellent. Capture.wav files were used with PESQM Abacus and WinEyeQ Traffic Capture to verify the voice QoS.

The AR routers can support a variety of different voice trunks which allow network administrators to select various types of configurations based on their needs. Some of the trunks supported are SIP, PRA, and ATO (FXO).

The AR routers also support wireless capabilities and 3G cards for mobile clients and added resiliency in a network deployment.

Test Bed Diagram



Source: Miercom, March 2012

How We Did It

The Huawei AR207V-P, AR1220VW, and AR3260 routers were fully evaluated for basic voice functionality, as well as voice trunking and additional services. Testing was conducted to verify that each of the features outlined in this report operated as described. Although we tested each individual router, the features in these models are supported on the other devices in the AR200, 1200, and 3200 series making the results applicable for all routers in the series.

Throughput was evaluated by sending two different traffic patterns through the router and using the traffic generator to measure the total throughput values.

Sections of this test required the use of a traffic generator to evaluate the features of the router. Two different traffic generators were used during the course of the testing, Ixia XM12 running IxNetwork version 5.50.121.48 and Spirent TestCenter running version 3.76.0076.

We used the Spirent Abacus 5000 to generate voice and SIP calls and to stress the voice portion of the router. The Abacus 5000 is capable of testing scalability, voice quality, and load handling.

Miercom recognizes Ixia (www.ixiacom.com) as an industry leader in energy efficiency testing of networking equipment. Ixia's unique approach utilizes coordination of energy measurements with network traffic load – allowing energy consumption to be graphed against network traffic volume. Real-world traffic is generated by Ixia's test platform and test applications, principally IxNetwork for Layer 2-3 routing and switching traffic and IxLoad for Layer 4-7 application traffic.

The tests in this report are intended to be reproducible for customers who wish to recreate them with the appropriate test and measurement equipment. Current or prospective customers interested in repeating these results may contact reviews@miercom.com for details on the configurations applied to the Device Under Test and test tools used in this evaluation. Miercom recommends customers conduct their own needs analysis study and test specifically for the expected environment for product deployment before making a product selection.

Miercom Performance Verified

The voice performance of Huawei AR G3 enterprise routers were verified by Miercom. In hands-on testing, Huawei demonstrated advanced v features such as:

- Support for a variety of voice trunks including SIP, PRA, and ATO
- Additional voice services including three-way conference calling and simultaneous/sequential ringing
- AR G3 routers had 100% SIP call completion without any dropped calls achieving 100% call completion
- Routers provide PBX voice functions, including conference calling and multi-language IVR

AR207V-P Router

AR1220VW Router

AR3260 Router

Huawei Technologies Co., Ltd.
http://enterprise.huawei.com

About Miercom's Product Testing Services

Miercom has hundreds of product-comparison analyses published over the years in leading network trade periodicals including Network World, Business Communications Review, Tech Web - NoJitter, Communications News, xchange, Internet Telephony and other leading publications. Miercom's reputation as the leading, independent product test center is unquestioned.

Miercom's private test services include competitive product analyses, as well as individual product evaluations. Miercom features comprehensive certification and test programs including: Certified Interoperable, Certified Reliable, Certified Secure and Certified Green. Products may also be evaluated under the NetWORKS As Advertised program, the industry's most thorough and trusted assessment for product usability and performance.

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