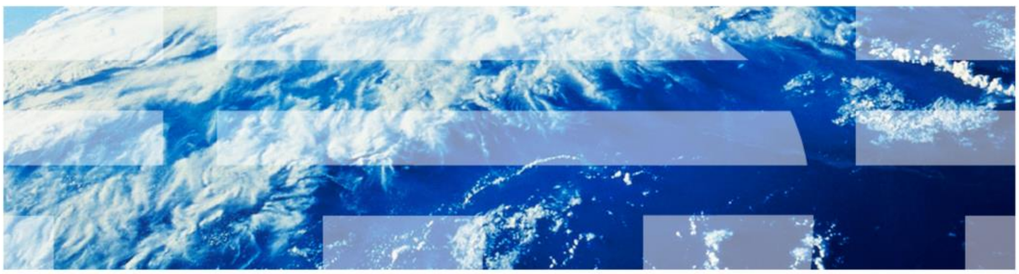
WebSphere Voice Response for AIX

Prerequisite considerations



This presentation will describe some of the basic prerequisites you must take into consideration when planning to setup a WebSphere® Voice Response system. While not completely necessary, a basic familiarity with the terms associated with telephony and interactive voice response systems will be helpful in understanding how to plan your system.

Goal

- Provide an overview of the steps necessary to plan a new WebSphere Voice Response version 6.1 system.

The goal of this presentation will be to provide a basic overview of the steps necessary when planning to setup a new WebSphere Voice Response version 6.1 system.

Areas of consideration

- Telephony network
- Server details
  - Hardware and software requirements
- Scalability

There are three main areas of consideration you must focus on when planning a new WebSphere Voice Response system: the telephony network, workstation details, and system scalability.

Telephony network (1 of 2)

- How will your telephony network interface with your voice processing system?
- Type of switch and how it will be configured?
  - Digital central office switch (exchange)
  - Private automated branch exchange (PABX)
  - Automatic call distributor (ACD) system
  - Host-controlled digital switch
- Type of digital trunk connection?
  - T-carrier level 1 (T1) or E-carrier level 1 (E1)

The first step you must consider when planning your WebSphere Voice Response system is how your telephony network will interface with your voice processing system. Since telephony regulations and public networks vary from country to country, WebSphere Voice Response provides a variety of connection facilities. You must consider what type of telephony switch you will use and how it will be configured. You must also decide if you will be using T1 or E1 cabling between your switch and your WebSphere Voice Response machine.

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Telephony network (2 of 2)

- Type of signaling protocol?
  - Channel associated signaling (CAS)
  - Common channel signaling (CCS)
    - Integrated Services Digital Network (ISDN)
    - Signaling System 7 (SS7)
- Estimate telephony traffic
  - Number of digital voice trunks and channels necessary
  - Formula to calculate this in the “General Information and Planning” guide

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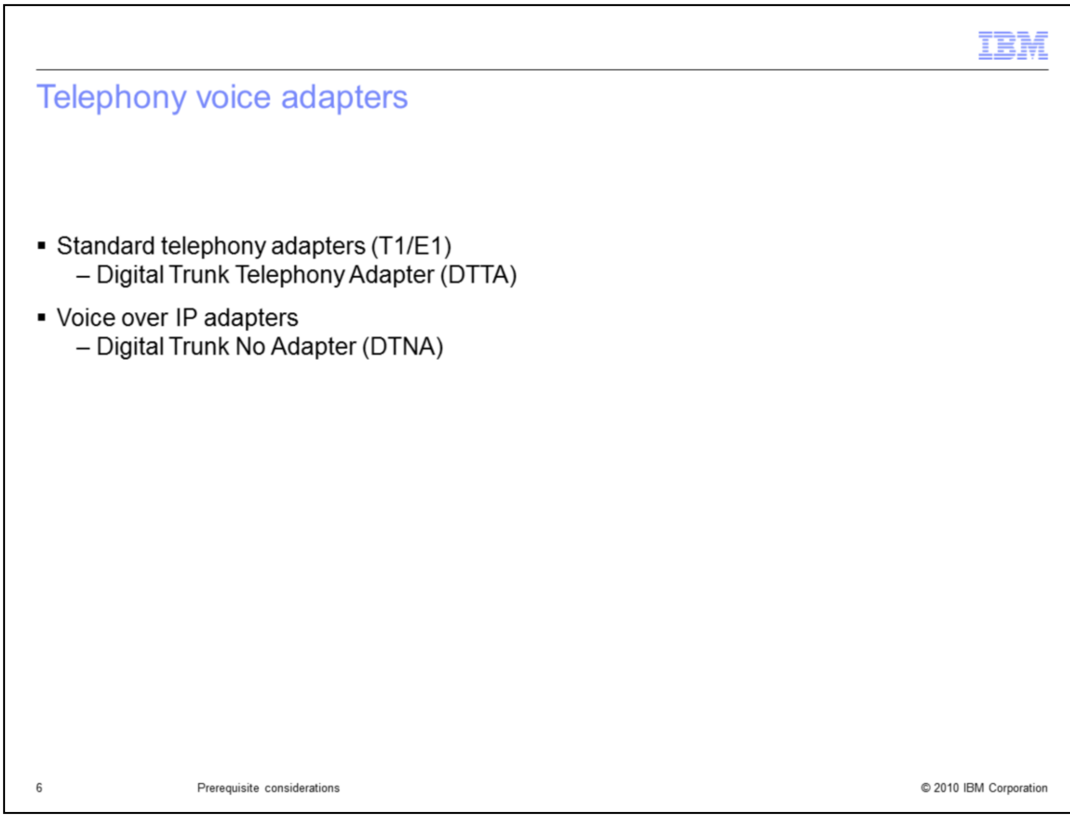
You must decide between two main types of signaling protocol: channel associated signaling or common channel signaling. Both protocol families have their benefits and restrictions, so you must weigh your options carefully.

With CAS, signaling information is carried in the voice channel or in a channel that is permanently tied to the voice channel. A number of different channel associated signaling protocols are available which each have their own set of telephony capabilities. Your choice is dependent on what subset of protocols your switch or PABX supports, and which protocols provide the functionality that your applications require. Although CAS is widely available in almost every country, there are a few disadvantages to selecting this protocol. For example, because only two or four bits of signaling information are available for each channel, the channel-associated signaling bits can be used only to pass very basic information between WebSphere Voice Response and the switch.

However, using common channel signaling can avoid all of the CAS shortcomings. This is because CCS uses fast messages sent bidirectionally down a single time slot of a trunk. This allows the switch and WebSphere Voice Response to communicate much more efficiently than they can with a CAS trunk. However, there are some disadvantages to CCS. First, the cost of implementing CCS can be more expensive. Second, if your signaling channel is lost you will lose functionality of all the voice channels controlled by that signaling channel.

It is also important to estimate your telephony traffic early on in the process of planning. Doing so will help you determine the number of WebSphere Voice Response machines and the number of channels necessary to handle incoming traffic. The number of voice channels will therefore determine the number of digital voice trunks and, in turn, the number of digital voice adapters necessary. Telephony traffic is measured in units called Erlangs. There is a basic Erlang formula to calculate this number. It is defined in the WebSphere Voice Response 4.2 “General Information and Planning” section of the WebSphere Voice Response information center.



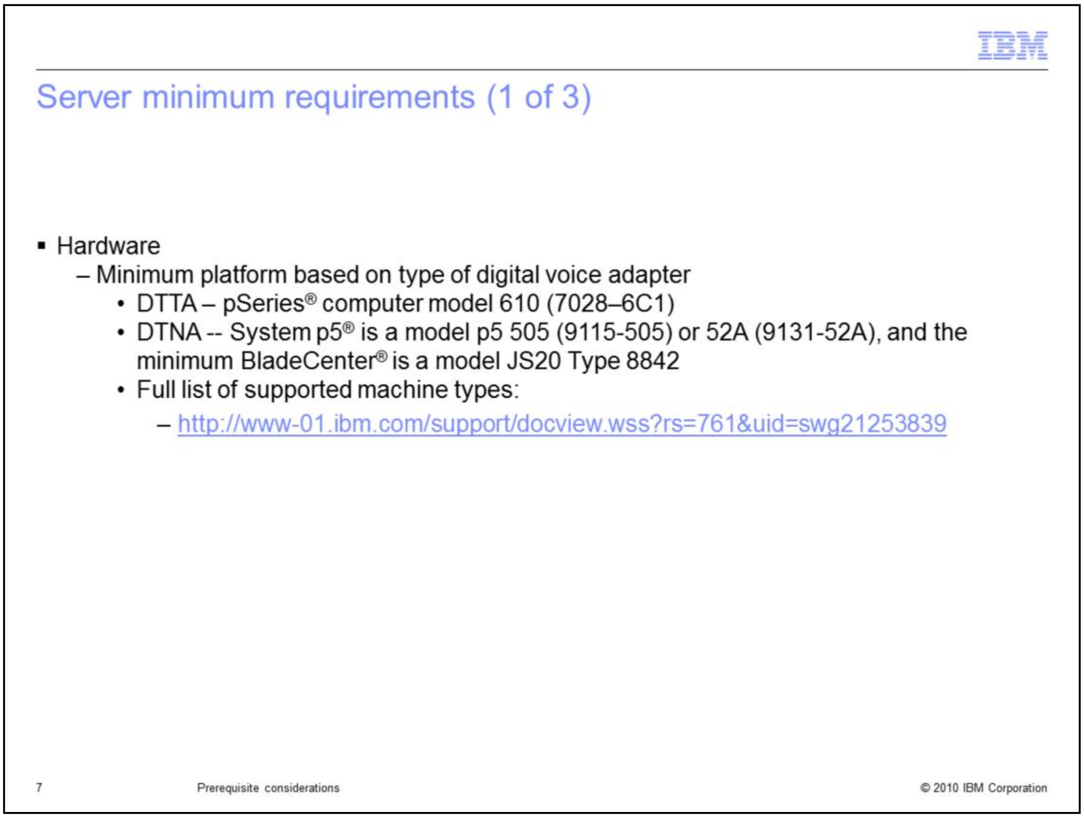
Currently, there are two types of telephony voice adapters supported for use with WebSphere Voice Response: DTTA, and DTNA.

When considering between a DTTA and a DTNA, these points come into play:

First, a DTTA setup may cost more since you would have to buy hardware adapters. Conversely, you may have to increase your server's RAM and processor since DTNA depends more on system resources due to the fact its software based.

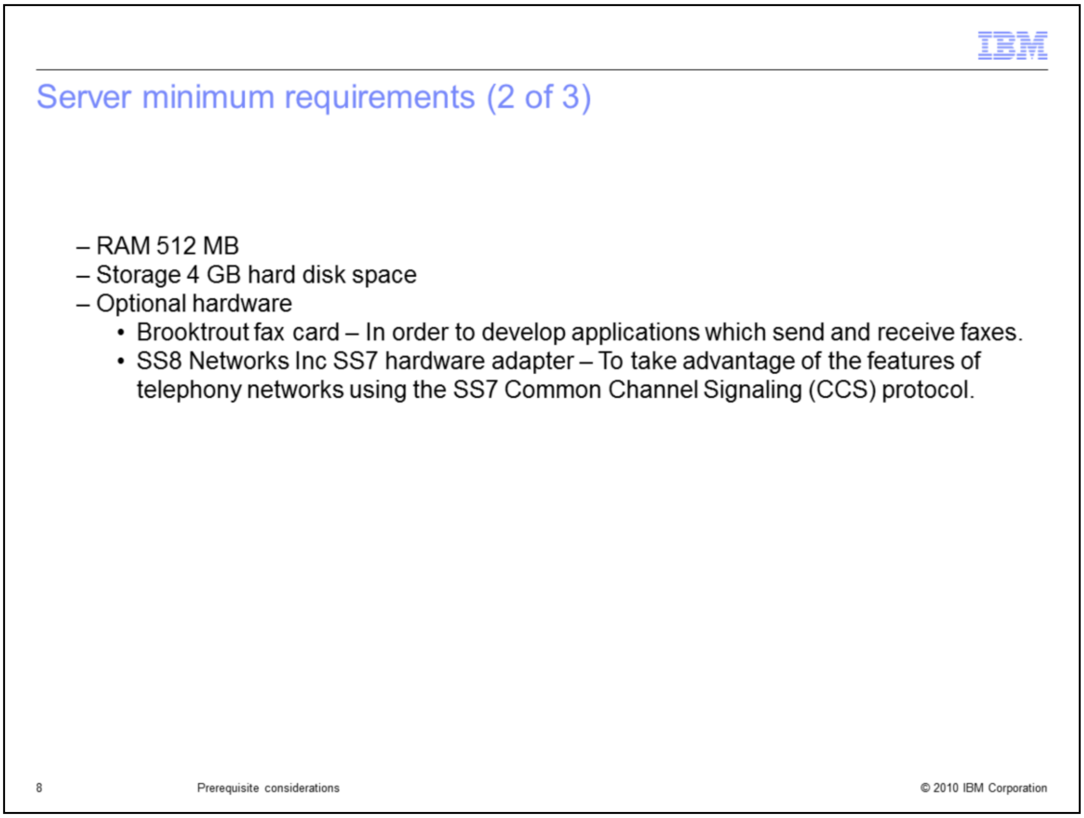
Second, DTNA will require a gateway unless you're only implementing VoIP calls. The gateway would be used to route standard telephony calls, such as landline or cellular, into the VoIP system. Similarly, a DTTA adapter would require a telephony switch to accept and route calls.

Third, the DTNA adapter does not support fax, echo cancellation, or tromboning between a T1 channel and a VoIP channel.



On this next slide, you will see the absolute minimum hardware requirements necessary to run a stand-alone WebSphere Voice Response system. WebSphere Voice Response must be run on a pSeries, system P5, or BladeCenter, depending on the type of digital voice adapter you will use. DTTA is a hardware based digital voice adapters and must be purchased separately from WebSphere Voice Response. DTNA is a new software based implementation of the previously available DTEA adapter. DTNA requires no additional hardware installation and uses VoIP by way of SIP which stands for Session Initiation Protocol.

The URL shown on this slide lists the currently supported machine types based on which voice adapter you're using.



A minimum of 512 MB of RAM is required in order to run only basic state table voice applications. With this minimum, you would also be limited to only 12 voice channels. Most production machines will want to have at least 2 to 8 GB of RAM in order to adequately handle high volumes of common voice applications. To further determine the amount of RAM you will need based on your voice application complexity, contact your IBM Sales Representative.

4 GB of hard disk space is required in order to install the various software components of WebSphere Voice Response in addition to its companion software. The amount of hard disk space necessary will increase based on the size and complexity of voice applications you create.

If you are planning on installing additional software on your WebSphere Voice Response server, such as voice recognition, you must plan for the increased resource requirements.

Server minimum requirements (3 of 3)

- Software prerequisites
  - Operating System
    - IBM AIX® V6.1 with minimum Technology Level 03 Service pack 01
    - WebSphere Voice Response supports a 64-bit AIX kernel
  - IBM DB2® WSE V9.5 (5765-F35) at minimum fix pack level 4. (provided with WebSphere Voice Response)
    - "limited use" copy, entitling customers to install and use DB2 WSE, subject to the terms and conditions of the license agreement that accompanies DB2 WSE
  - IBM XL C Enterprise Edition for AIX, V9 (5724-S70)
    - the RTE component included with the AIX installation media.

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WebSphere Voice Response must be run on IBM's AIX operating system. WebSphere Voice Response will run in either a 32-bit or 64-bit AIX kernel. There is no advantage to either. You only need to run in a 64-bit AIX kernel if you have other applications which require it.

WebSphere Voice Response ships with DB2 Workgroup Server Edition version 9.5. DB2 is provided only for the storage and management of data used by WebSphere Voice Response, and if it is to be used by other applications, a separate license must be purchased.

Optional software

- Possible additional software based on expected application use
  - IBM 32-bit SDK for AIX, Java™ Technology Edition, V6, Service Refresh 3 (SR3), level 6.0.0.75
    - If running VXML, CCXML or Java API applications
  - IBM XL C/C++ Enterprise Edition for AIX, V9 compiler (5724-S71)
    - If developing Custom Servers
  - AIX Communications Server for AIX 6.3
    - If using 3270 communications
  - IBM Developer Kit and Runtime for AIX, Java Technology Edition (JDK) 1.4.1 or 1.5
    - If using CCXML, VoiceXML or Java voice applications

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Depending on the type of voice applications you want to run or develop, you may have to install additional software as listed on this slide.

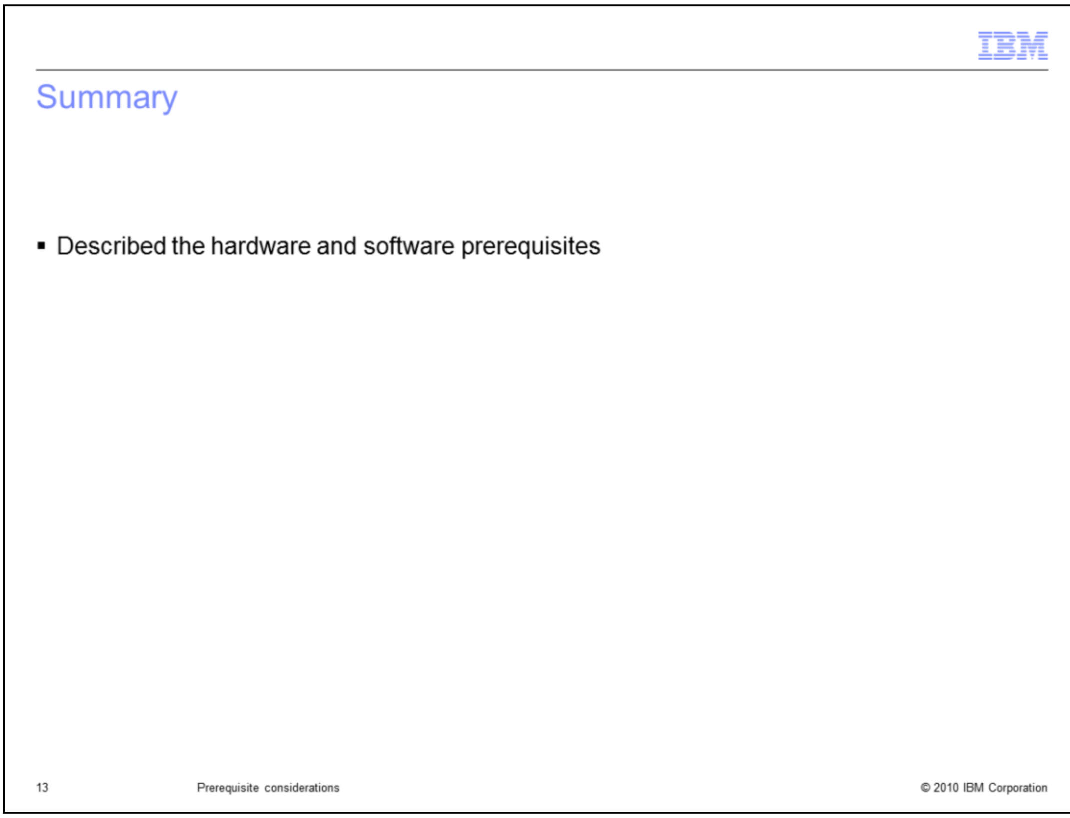
Scalability

- Single system image (SSI)
  - Local area network used to connect a cluster of WebSphere Voice Response systems.
  - Can share all the application data (state tables, custom servers) and all the voice data (voice segments, voice messages).
- IBM's HACMP™ (High Availability Cluster Multi-Processing) software
  - Reduces the chances of your business being stopped at a single point of failure.
  - Handles shared external disks between nodes so that when a failure is detected, HACMP allows your network to continue working by using different paths to data and applications.

The scalability of WebSphere Voice Response can be manipulated by creating a single system image, or SSI cluster. By connecting a group of WebSphere Voice Response systems, the cluster can share all the application and voice data.

In addition to improving the scale of WebSphere Voice Response by setting up a SSI cluster, you can use IBM's HACMP software to reduce the chances of your business being stopped at a single point of failure. HACMP handles shared external disks between nodes so that a failure will not bring down your entire environment.

Further detailed information on the prerequisites can be found in the "General Information and Planning" and the "Installation" sections of the WebSphere Voice Response Information Center at the first URL listed, while the latest hardware and software requirements can be found at the second URL listed.



In summary, you should now have a basic understanding of the logistical considerations in addition to the hardware and software prerequisites necessary for setting up a new WebSphere Voice Response system.

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