PBX Server Configuration

01/09/2024 6:46 pm EST

Prerequisites

The following components are necessary for the PBXServer installation:

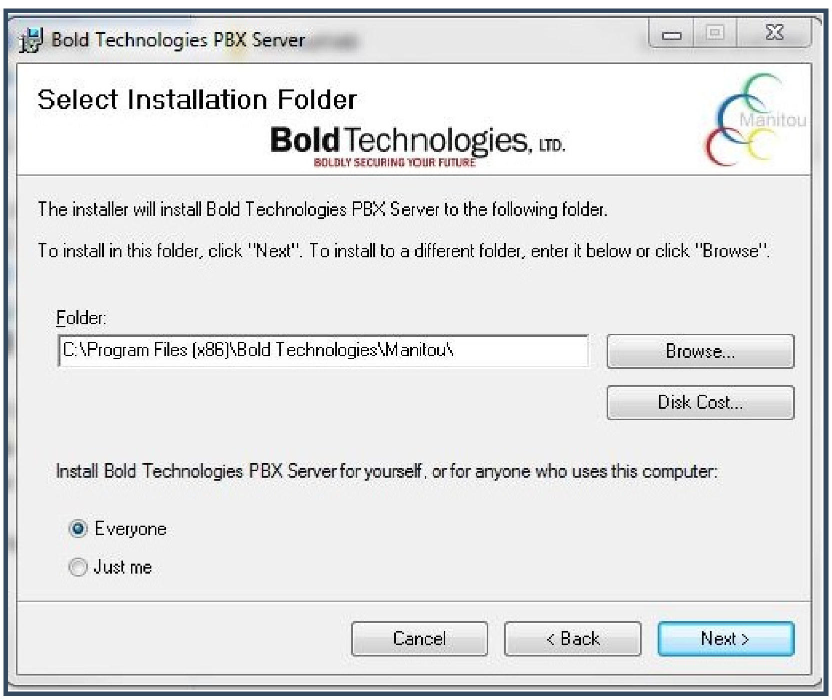
- Windows Server 2008 R2 or higher
- Microsoft Speech API
- BoldPBX.tsp TAPI Driver installed on the PBX Server and every workstation that will use the PBX AutoDialer (the 32-bit version is contained in Package 7, and the 64-bit version must be installed manually)
- Proper licensing for PBX Server Automation
  - Include Audio Accounts Manitou Licensing
- Proper licensing for DMCC (Avaya - necessary phone and media are covered by this one license)
- Proper licensing for TSAPI (Avaya)
- Proper licensing for the PBX AutoDialer
  - Note: the number of licenses necessary for the PBX AutoDialer corresponds to the number of concurrent
- AutoDial sessions that can be active in the PBXServer.
- Avaya Phone Switch Hardware (necessary for PBX Standard only)
- Tadiran Phone Switch Hardware (necessary for PBX Standard only)
- Port 2555 forwarded from PBX Server to Aeonix Gateway (necessary for PBX Standard only)

Installing the PBX Server

Perform the following steps to extract the PBXServer installation files:

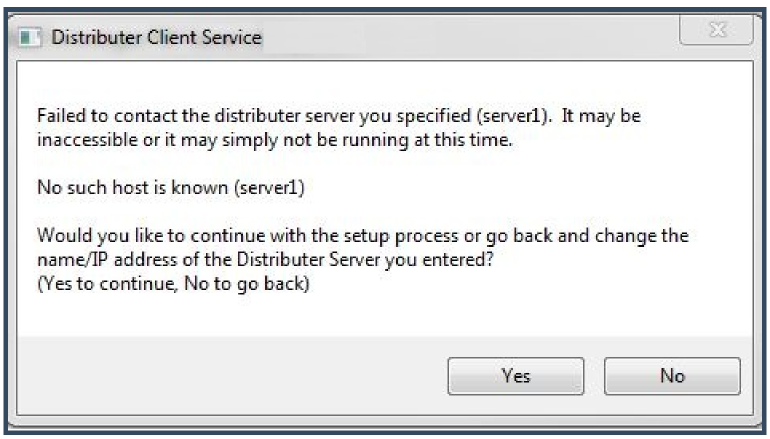
1. Copy the PBXServer_Setup.exe file from the install package onto the desktop of the machine onto which you want to install it. If this will be used with TwoWay it typically will be installed on the MediaGateway Server(s).
2. Right-click on PBXServer_setup.exe, and then select the Run as Administrator option.
?Result: the Bold Technologies PBX Server window displays as shown in the following screenshot:

3. Click Next.
Result: the following Bold Technologies PBX Server window displays:

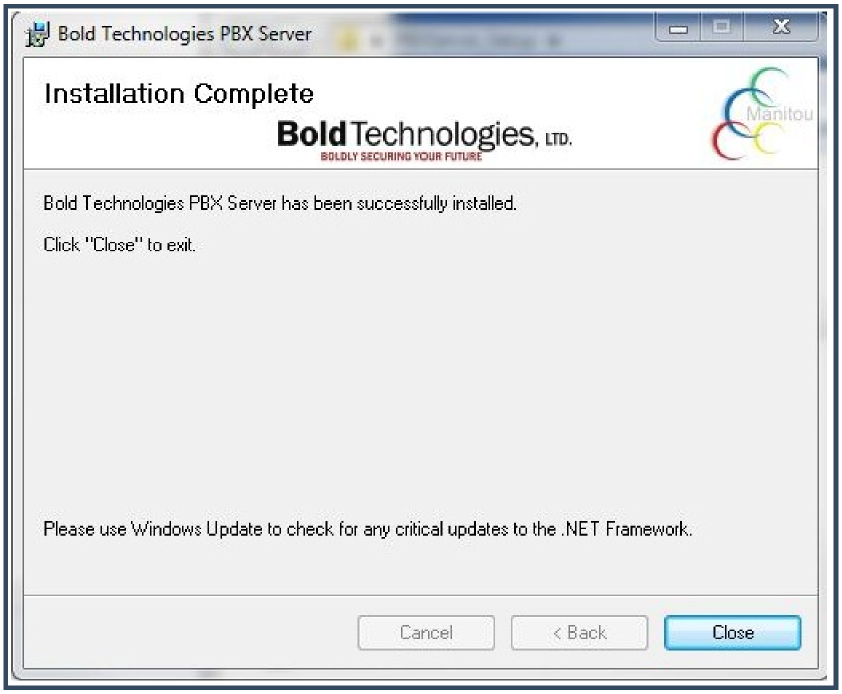


4. Select a destination for the PBXServer files.
5. If you want the PBXServer to be available for anyone who uses the Workstation, select the Everyone option. If you want the PBXServer to only be available to you, select Just Me.
6. Click Next.
Result: the following License Agreement window displays:

7. Select the I Agree option and then click Next and type in the Name of the distributer server.
Result: the following Distributer Client Service window might display:



8. If the Distributer Client Service window is displayed after you performed the previous step, click Yes.
Result: the following Installation Complete window displays:



9. Click Close.

Updating the PBX Server with the Distributor

1. Push out the necessary packages for the MediaGateway and send out package 200

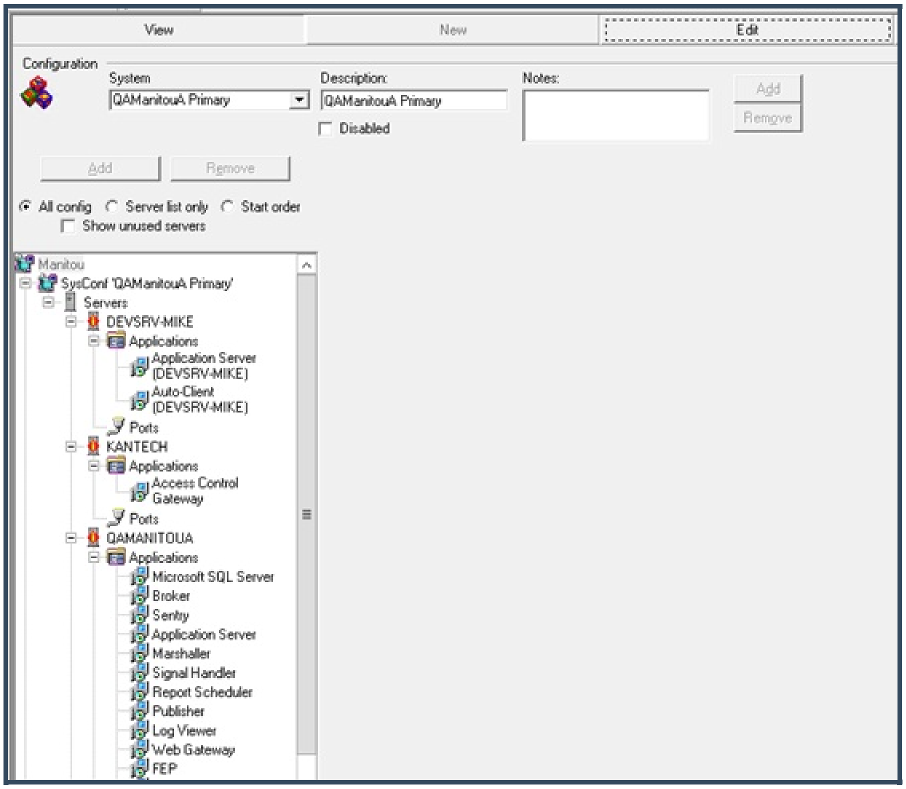
Setting up the PBX Server in the Supervisor Workstation Configuration Form

You must add the PBX Server to your Manitou configuration in the Supervisor Workstation.

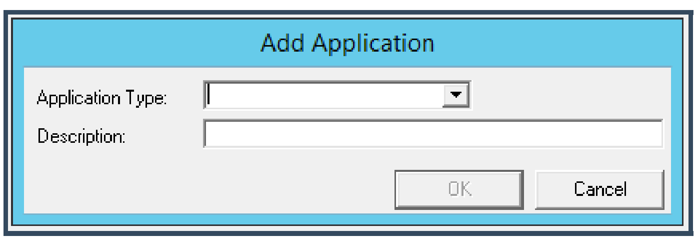
Perform the following steps to add PBXServer as a Manitou application:

1. Open the Manitou Supervisor Workstation.
2. Navigate to the Maintenance menu, and then select Setup and Configuration.

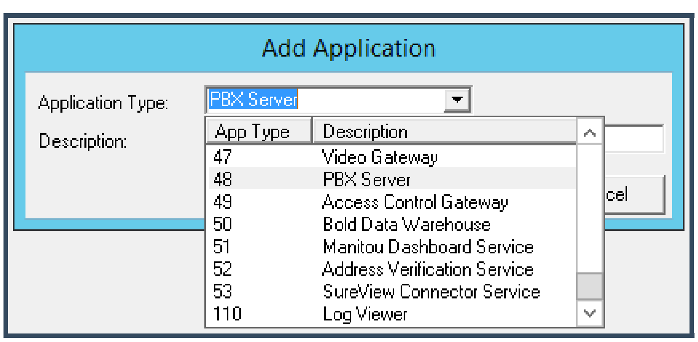
Result: the Configuration form displays as shown in the following screenshot:



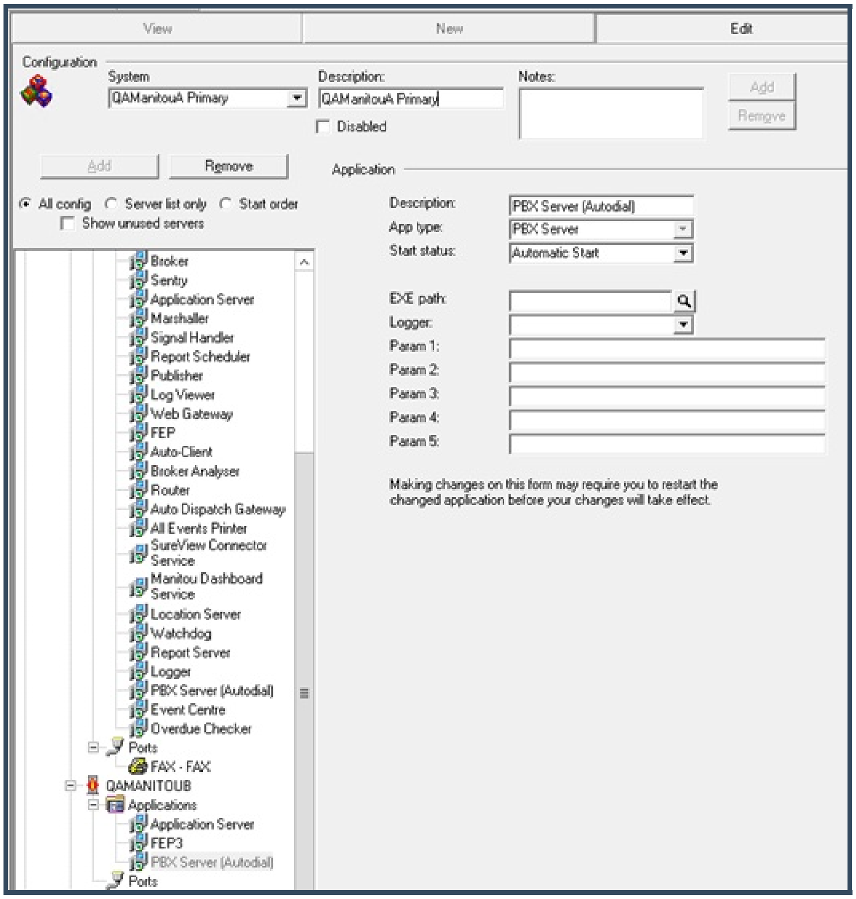
3. Locate the server onto which you want to install PBXServer in the Navigation Tree.
4. Select the "Applications" icon located under the icon for the server onto which you want to install PBXServer.
5. Click Edit, and then click Add.
Result: the Add Application window displays as shown in the following screenshot:



6. Select 48 PBX Server from the Application Type drop-down list box as shown in the following screenshot:



7. Enter a description into the Description field, and then click OK.
Result: the PBX Server you added as an application now displays on the Configuration form as shown in the following screenshot:



8. Click Save.

PBXVoiceConfigurations.config

Server 2016 does not include Microsoft Anna, the information below will show you how to set up the configuration to support Microsoft Zira.

This configuration designates to PBXServer which voice to use when creating Text-To-Speech. The “TTSConfiguration” MFC in the MediaGateway 2 allows you to define the voice you want to use in the menu. This is useful for certain extensions for which you want to use a Spanish-speaking voice. See http://msdn.microsoft.com/en-us/library/ms717077%28v=vs.85%29.aspx for more information.

The following screenshot displays potential configuration settings for the “PBXVoiceConfigurations.config” file:

```xml
<?xml version="1.0" encoding="utf-8"?>
<VoiceConfigurations xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xmlns:xsd="http://www.w3.org/2001/XMLSchema">
  <VoiceConfigurationList>
    <VoiceConfiguration>
      <OptionalAttributes></OptionalAttributes>
      <RequiredAttributes>Language=409</RequiredAttributes>
      <ConfigurationId>1</ConfigurationId>
      <Header>&lt;pitch absmiddle="4"/&gt;</Header>
      <Footer>something else=</Footer>
      <Rate>0</Rate>
      <Volumn>100</Volumn>
    </VoiceConfiguration>
    <VoiceConfiguration>
      <OptionalAttributes>Name=Anna</OptionalAttributes>
      <RequiredAttributes />
      <ConfigurationId>2</ConfigurationId>
      <Header></Header>
      <Footer></Footer>
      <Rate>0</Rate>
      <Volumn>100</Volumn>
    </VoiceConfiguration>
    <VoiceConfiguration>
      <OptionalAttributes></OptionalAttributes>
      <RequiredAttributes>Language=409; Name=Microsoft Zira Desktop</RequiredAttributes>
      <ConfigurationId>0</ConfigurationId>
      <Header></Header>
      <Footer></Footer>
      <Rate>0</Rate>
      <Volumn>100</Volumn>
    </VoiceConfiguration>
  </VoiceConfigurationList>
</VoiceConfigurations>
```

```xml
<?xml version="1.0" encoding="utf-8"?>
<VoiceConfigurations xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xmlns:xsd="http://www.w3.org/2001/XMLSchema">
  <VoiceConfigurationList>
    <VoiceConfiguration>
      <OptionalAttributes>Vendor=Loquendo</OptionalAttributes>
      <RequiredAttributes>Language=409</RequiredAttributes>
      <ConfigurationId>1</ConfigurationId>
      <Header>&lt;pitch absmiddle="4"/&gt;</Header>
      <Footer>something else=</Footer>
      <Rate>0</Rate>
      <Volumn>100</Volumn>
    </VoiceConfiguration>
    <VoiceConfiguration>
      <OptionalAttributes>Name=Anna</OptionalAttributes>
      <RequiredAttributes />
      <ConfigurationId>2</ConfigurationId>
      <Header></Header>
      <Footer></Footer>
      <Rate>0</Rate>
      <Volumn>100</Volumn>
    </VoiceConfiguration>
 <VoiceConfiguration>
      <OptionalAttributes></OptionalAttributes>
      <RequiredAttributes>Language=409; Name=Microsoft Zira Desktop</RequiredAttributes>
      <ConfigurationId>0</ConfigurationId>
      <Header></Header>
      <Footer></Footer>
      <Rate>0</Rate>
      <Volumn>100</Volumn>
    </VoiceConfiguration>
  </VoiceConfigurationList>
</VoiceConfigurations>
```

PBXServer.config

The following screenshot displays potential configuration settings for the “PBXServer.config” file:?

```xml
<?xml version="1.0" encoding="utf-8"?>
<PBXServerConfig xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance" xmlns:xsd="http://www.w3.org/2001/XMLSchema">
    <Port>6110</Port>
    <BrokerHost>localhost</BrokerHost>
    <LoggerHost>local_logging_only</LoggerHost>
    <DnisLength>4</DnisLength>
    <InputTerminationCharacter>35</InputTerminationCharacter>
    <HangUpCallIfAlreadyConnectedForSessionInit>false</HangUpCallIfAlreadyConnectedForSessionInit>
    <MaxDtmfDigits>31</MaxDtmfDigits>
    <VoicesPath>c:\Media</VoicesPath>
    <RecordPath>c:\Media</RecordPath>
    <TextToSpeechThreads>3</TextToSpeechThreads>
    <TtsTimeout>10000</TtsTimeout>
    <ScpWaveFiles>true</ScpWaveFiles>
    <ScpServer>172.16.143.999</ScpServer>
    <ScpDirectory>/home/aeonixadmin/aeonix/resources/audio/</ScpDirectory>
    <ScpUser>aeonixadmin</ScpUser>
    <ScpPassword>anx</ScpPassword>
    <VoiceToUse>0</VoiceToUse>
    <ConnectingToPbxSimulator>false</ConnectingToPbxSimulator>
    <PlayTimeInMsIfCantDeterminePlayTime>5000</PlayTimeInMsIfCantDeterminePlayTime>
    <PauseInMsBeforeSendingDnisAndAniToReceiver>400</PauseInMsBeforeSendingDnisAndAniToReceiver>
    <DnisAndAniToneDurationInMs>80</DnisAndAniToneDurationInMs>
    <DnisAndAniBetweenToneDurationInMs>80</DnisAndAniBetweenToneDurationInMs>
    <DnisAndAniFormat>A#D#</DnisAndAniFormat>
    <LoggerLevelOverride>5</LoggerLevelOverride>
    <DriverToLoad>PbxServer.Driver.Aeonix.dll</DriverToLoad>
    <ScpDeleteArguments>"rm -f '%ScpDirectory%%BaseFileName%'"</ScpDeleteArguments>
    <TtsFilesToCreate>
        <TtsConfig>
            <FileExtension>.ulaw</FileExtension>
            <SpeechAudioFormatType>49</SpeechAudioFormatType>
        </TtsConfig>
        <!--<TtsConfig>
            <FileExtension>.alaw</FileExtension>
            <SpeechAudioFormatType>41</SpeechAudioFormatType>
        </TtsConfig>
        <TtsConfig>
            <FileExtension>.g729</FileExtension>
            <SpeechAudioFormatType>49</SpeechAudioFormatType>
        </TtsConfig>-->
    </TtsFilesToCreate>
    <DtmfToneDelayPercentage>1.25</DtmfToneDelayPercentage>
    <DtmfPauseDelayPercentage>1.25</DtmfPauseDelayPercentage>
    <HowLongToDndReceiverExtensionInSeconds>120</HowLongToDndReceiverExtensionInSeconds>
    <LogAnswerPendingResponses>false</LogAnswerPendingResponses>
</PBXServerConfig>
```

The following options are available for the “PBXServer.config” file:

Port –the port number on which you want the PBXServer to listen. This is the port you should configure in the MediaGateway 2 when setting up the PBXServer connection.

```
<Port>6110</Port>
```

BrokerHost – the Broker machine name to which you want the PBXServer to connect. PBXServer uses this only when it is not passed the name as a parameter.

```
<BrokerHost>localhost</BrokerHost>
```

LoggerHost – the Logger machine name to which you want the PBXServer to connect. PBXServer uses this only when it is not passed a name as a parameter. If you set this to “local_logging_only” and you have the “BOLD_FILE_LOG” environmental variable set to log PBXServer logs, it will only log to the file and not to the Logger.

```
<LoggerHost>localhost</LoggerHost>
```

DnisLength – this indicates the number of characters you want to display for DNIS (for e.g., users might receive five characters from a Receiver, but only want to use four).

```
<DnisLength>7</DnisLength>
```

InputTerminationCharacter – this indicates the input character you want the user to press to indicate he is now finished with the preceding command (for e.g., “Enter your user ID followed by the pound sign).

```
<InputTerminationCharacter>35</InputTerminationCharacter>
```

Hangupcallifalreadyconnectedforsessioninit – this indicates whether the system should automatically terminate the call if it is already connected to a monitored extension.

```
<HangUpCallIfAlreadyConnectedForSessionInit>false</HangUpCallIfAlreadyConnectedForSessionInit>
```

Maxdtmfdigits – this indicates the maximum number of digits allowed when a user is entering a user ID or password

```
<MaxDtmfDigits>31</MaxDtmfDigits>
```

Voicespath – this designates the path to store created text-to-speech files.

```
<VoicesPath>c:\Media</VoicesPath>
```

Recordpath – this designates the path to store voice recordings.

```
<RecordPath>c:\Media</RecordPath>
```

Texttospeechthreads – this indicates how many independent processes the user wants to handle text-to-speech operations.

```
<TextToSpeechThreads>5</TextToSpeechThreads>
```

Ttstimeout – this indicates the maximum number of milliseconds the text-to-speech processor should wait for a text-to-speech file to be generated.

```
<TtsTimeout>10000</TtsTimeout>
```

SCPWaveFiles – this indicates whether the user wants the system to copy text-to-speech .wav files created to another server. This setting is required for both the Avaya and Aeonix integrations.

```
<ScpWaveFiles>true</ScpWaveFiles>
```

Scpserver – this indicates the server to which you want the system to copy text-to-speech .wav files.

Scpdirectory – this specifies the directory on the server to which you want the system to copy text-to-speech .wav files.

```
<ScpDirectory>/home/aeonixadmin/aeonix/resources/audio/</ScpDirectory>
```

Scpuser – this indicates a user login.

```
<ScpUser>aeonixadmin</ScpUser>
```

Scp password – this indicates a user password.

```
<ScpPassword>anx</ScpPassword>
```

Voicetouse –this option references the following configuration file: “pbxvoicesconfigurations.config”. The “pbxvoicesconfigurations.config” file determines which configuration to use from the “PBXServer.config” file. Set this to the voice you want to use for PBX.

```
<VoiceToUse>0</VoiceToUse>
```

Connectingtopbxsimulator – this internal flag determines whether the user wants to connect to the PBX Simulator rather than to the actual server. This option should always be set to false.

```
<ConnectingToPbxSimulator>false</ConnectingToPbxSimulator>
```

Playtimeinmsifcantdetermineplaytime – this option indicates the number of milliseconds you want the system to wait before timing out if the duration of a .wav file cannot be determined.

```
<PlayTimeInMsIfCantDeterminePlayTime>5000</PlayTimeInMsIfCantDeterminePlayTime>
```

Pauseinmsbeforesendingdnisandanitoreceiver – this option indicates the number of milliseconds you want the system to pause before sending DNIS and ANI information to the Receiver. “400” is the standard value for the Osborne Hoffman 2000E Receiver. This setting is unnecessary if you are connecting to an Aeonix server.

```
<PauseInMsBeforeSendingDnisAndAniToReceiver>400</PauseInMsBeforeSendingDnisAndAniToReceiver>
```

Dnisandanitonedurationinms – this option indicates the duration (in milliseconds) you want the system to send DTMF digits to the Receiver. “80” is the standard value. This setting is unnecessary if you are connecting to an Aeonix server.

```
<DnisAndAniToneDurationInMs>80</DnisAndAniToneDurationInMs>
```

Dnisandaniformat – this option indicates two tags that are replaced. “A” is replaced with the ANI, and “D” is replaced with the DNIS. This is a required setting for the Osborne Hoffman 2000E Receiver. This setting is unnecessary if you are connecting to an Aeonix server.

```
<DnisAndAniFormat>A#D#</DnisAndAniFormat>
```

Dnisandanibetweentonedurationinms – this option indicates the number of milliseconds you want the system to pause between each DNIS or ANI digit it sends to the Receiver. This setting is unnecessary if you are connecting to an Aeonix server.

```
<DnisAndAniBetweenToneDurationInMs>80</DnisAndAniBetweenToneDurationInMs>
```

Loggerleveloverride – this option indicates how detailed you want Logger information to display. This overrides the Broker Log Level setting.

```
<LoggerLevelOverride>5</LoggerLevelOverride>
```

Drivertoload – this option specifies the type of phone switch to which you want to connect. For Avaya, the switch to designate is "PbxServer.Driver.Avaya.dll" For Aeonix, the switch to designate is "PBXServer.Driver.Aeonix.dll".

```
<DriverToLoad>PbxServer.Driver.Aeonix.dll</DriverToLoad>
```

ScpDeleteArguments – this option indicates how you want a user to delete the file using a remote "SSH" command.

```
<ScpDeleteArguments>"rm -f '%ScpDirectory%%BaseFileName%'"</ScpDeleteArguments>
```

TtsFilesToCreate – this option specifies to the PBXServer how many .wav files to create and which type they should be.

```
<TtsFilesToCreate>
 <TtsConfig>
  <FileExtension>.ulaw</FileExtension>
  <SpeechAudioFormatType>49</SpeechAudioFormatType>
 </TtsConfig>
 <!--<TtsConfig>
  <FileExtension>.alaw</FileExtension>
  <SpeechAudioFormatType>41</SpeechAudioFormatType>
 </TtsConfig>
 <TtsConfig>
  <FileExtension>.g729</FileExtension>
  <SpeechAudioFormatType>49</SpeechAudioFormatType>
 </TtsConfig>-->
</TtsFilesToCreate>
```

FileExtension – this option indicates the file extension to the PBXServer.

```
  <FileExtension>.ulaw</FileExtension>
```

SpeechAudioFormatType – this option indicates to the PBXServer the audio file type. See Appendix A for a list of available audio types.

```
  <SpeechAudioFormatType>49</SpeechAudioFormatType>
```

DtmfToneDelayPercentage – this option indicates to the PBXServer how much variance you want in the duration of DTMF tones. For example, a user wants a PBXServer DTMF tone to play for 100 milliseconds. If it takes 125 milliseconds to play the tone, this value should be set to 1.25.

```
<DtmfToneDelayPercentage>1.25</DtmfToneDelayPercentage>
```

DtmfPauseDelayPercentage – this option indicates to the PBXServer how much variance you want in the silent duration between DTMF tones.

```
<DtmfPauseDelayPercentage>1.25</DtmfPauseDelayPercentage>
```

HowLongToDndReceiverExtensionInSeconds – this option indicates the maximum “Do Not Disturb” duration in seconds before the Receiver extension times out. As of the publication of this document, this option is only available for Aeonix drivers.

```
<HowLongToDndReceiverExtensionInSeconds>120</HowLongToDndReceiverExtensionInSeconds>
```

LogAnswerPendingResponses – this option indicates whether the system should log pending “Answer” responses. If set to true, it substantially increases log size. This option is typically set to false.

Appendix A: Audio Types

This list indicates available audio types for the PBXServer:
[TypeLibVar(64)]
SAFTDefault = -1,
[TypeLibVar(64)]
SAFTNoAssignedFormat = 0,
[TypeLibVar(64)]
SAFTText = 1,
[TypeLibVar(64)]
SAFTNonStandardFormat = 2,
[TypeLibVar(64)]
SAFTExtendedAudioFormat = 3,
SAFT8kHz8BitMono = 4,
SAFT8kHz8BitStereo = 5,
SAFT8kHz16BitMono = 6,
SAFT8kHz16BitStereo = 7,
SAFT11kHz8BitMono = 8,
SAFT11kHz8BitStereo = 9,
SAFT11kHz16BitMono = 10,
SAFT11kHz16BitStereo = 11,
SAFT12kHz8BitMono = 12,
SAFT12kHz8BitStereo = 13,
SAFT12kHz16BitMono = 14,
SAFT12kHz16BitStereo = 15,
SAFT16kHz8BitMono = 16,
SAFT16kHz8BitStereo = 17,
SAFT16kHz16BitMono = 18,
SAFT16kHz16BitStereo = 19,
SAFT22kHz8BitMono = 20,
SAFT22kHz8BitStereo = 21,
SAFT22kHz16BitMono = 22,
SAFT22kHz16BitStereo = 23,
SAFT24kHz8BitMono = 24,
SAFT24kHz8BitStereo = 25,

SAFT24kHz16BitMono = 26,
SAFT24kHz16BitStereo = 27,
SAFT32kHz8BitMono = 28,
SAFT32kHz8BitStereo = 29,
SAFT32kHz16BitMono = 30,
SAFT32kHz16BitStereo = 31,
SAFT44kHz8BitMono = 32,
SAFT44kHz8BitStereo = 33,
SAFT44kHz16BitMono = 34,
SAFT44kHz16BitStereo = 35,
SAFT48kHz8BitMono = 36,
SAFT48kHz8BitStereo = 37,
SAFT48kHz16BitMono = 38,
SAFT48kHz16BitStereo = 39,
SAFTTrueSpeech_8kHz1BitMono = 40,
SAFTCCITT_ALaw_8kHzMono = 41,
SAFTCCITT_ALaw_8kHzStereo = 42,
SAFTCCITT_ALaw_11kHzMono = 43,
SAFTCCITT_ALaw_11kHzStereo = 44,
SAFTCCITT_ALaw_22kHzMono = 45,
SAFTCCITT_ALaw_22kHzStereo = 46,
SAFTCCITT_ALaw_44kHzMono = 47,
SAFTCCITT_ALaw_44kHzStereo = 48,
SAFTCCITT_uLaw_8kHzMono = 49,
SAFTCCITT_uLaw_8kHzStereo = 50,
SAFTCCITT_uLaw_11kHzMono = 51,
SAFTCCITT_uLaw_11kHzStereo = 52,
SAFTCCITT_uLaw_22kHzMono = 53,
SAFTCCITT_uLaw_22kHzStereo = 54,
SAFTCCITT_uLaw_44kHzMono = 55,
SAFTCCITT_uLaw_44kHzStereo = 56,
SAFTADPCM_8kHzMono = 57,
SAFTADPCM_8kHzStereo = 58,
SAFTADPCM_11kHzMono = 59,
SAFTADPCM_11kHzStereo = 60,
SAFTADPCM_22kHzMono = 61,
SAFTADPCM_22kHzStereo = 62,
SAFTADPCM_44kHzMono = 63,
SAFTADPCM_44kHzStereo = 64,
SAFTGSM610_8kHzMono = 65,
SAFTGSM610_11kHzMono = 66,
SAFTGSM610_22kHzMono = 67,
SAFTGSM610_44kHzMono = 68,