PBXpress Admin Guide

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PBXpress Admin Guide
V.2.1.4 October 2008

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Preface

This document provides general information about PBXpress VoIP PBX, as well as installation and configuration instructions.

Where to Get the Latest Version of This Guide

The online copy of this guide is always up-to-date, integrating the latest changes to the product. You can access the latest copy of this guide at: **www.pbxpress.com/docs**.

Conventions

This publication uses the following conventions:

- Commands and keywords are given in **boldface**
- Terminal sessions, console screens, or system file names are displayed in `fixed width font`

Caution indicates that the described action might result in program malfunction or data loss.

NOTE: Notes contain helpful suggestions about or references to materials not contained in this manual.

Timesaver means that you can save time by performing the action described in the paragraph.

Tips provide information that might help you solve a problem.

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Introduction

Let's answer a few simple questions before we proceed.

What is PBX?

PBX (Private Branch Exchange) is basically a phone switch for office organization. It connects phones inside your office with the rest of the world, allowing you to reach John just by dialing 501, and so on. This is that strange box sitting somewhere in a dark room, the one with all the cables connected to it, which most people in your organization have no idea how to work with.

What is PBXpress?

PBXpress is a modern-day replacement for traditional PBX systems. Instead of working with wires and cables, it utilizes Internet technologies. In particular, it uses VoIP (voice over IP) technology to transmit calls.

What are the benefits?

- Reduced hardware costs. PBXpress runs on a standard PC platform, which is much cheaper than specialized PBX equipment.
- Easy deployment and scalability. With traditional PBX, you need to have a physical wire from the PBX box to the phone on your desk. What if, during building construction, only one pair of cables was installed in your room, and you now need 3 phone lines? What if four of your employees are located not in your main office building, but in a sales center several blocks away? What if you want to use your phone while on a business trip? With IP-based PBXpress, all of this is no problem. You just need network connectivity to be able to use the phone service.
- Huge cost savings on long-distance and international phone calls. With PBXpress, you can utilize any of the many available VoIP carriers and terminate calls to them. In just one minute, you can start forwarding calls that cost only $0.05/min to a Moscow-based company, while still retaining Packet8 as the carrier for most of your international destinations, and Qwest as your provider for local calls.
- Easy to manage. No need to do PBX programming using a console interface and cryptic language from the manual – everything is web-based.
- Ability to record all phone conversations. Worried about who made all those calls to the Dominican Republic? Wish you had some record of your previous verbal agreement with a vendor? Want to analyze whether your new salesperson does the right thing on the phone? No problem: PBXpress can record every

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conversation made through it, storing them on CD-ROM or DVD-ROM in a format which can be played back on any computer.

- Integrated voicemail and auto-attendant, music on hold.
- ACD – automatic call distribution queues.
- Solution for networks with firewall/NAT. You've tried to use a residential VoIP company's service in your office, but couldn't get any audio? The most likely reason for this is that your firewall is too strict, and does not allow network traffic to pass. PBXpress can act as a bridge across the firewall for VoIP calls, eliminating such problems.
- Support for cluster and multi-office deployments
- Integration with an external billing system via RADIUS. PBXpress supports standard Cisco VSA attributes for VoIP calls, so it can send call information to an external billing solution (for example, PortaBilling100 – **portaone.com/products/portabilling/pb100/**) for further call processing/rating.

What's New in the Latest Release?

PBXpress is a modern-day replacement for traditional PBX systems. Instead of wires and cables, it makes use of Internet technologies such as VoIP (Voice over IP) to transmit calls. Features include:

- Operator console for on-screen handling of current calls;
- Ability to upload user picture into extension information;
- Admin groups for subsetting the IP PBX environment;
- Support for smart UPS.

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Usage Scenarios

PSTN Call Fail-over

Internet reliability is a very important issue when using VoIP. PBXpress offers a risk-free approach to this problem. PBXpress has an interface to your local PSTN line, so if an outgoing call fails via IP, it will be automatically routed via your existing PSTN line. This is transparent for your end users, so the worst-case scenario (no Internet connectivity available for VoIP calls at all) when using PBXpress will be exactly what you have right now, with a local telco used for outgoing calls.

Network Bridging

You have an office network and a firewall which protects it. Unfortunately, it looks as if the firewall blocks VoIP UDP traffic, so when you try to call someone you hear one-way audio, and people cannot call you at all. Currently you are unable to change your firewall configuration (this is either not possible, e.g. the firewall is built into your router, or your network administrator is unsure how to do it, or there is some other reason).
PBXpress offers an easy solution to this problem: two network interfaces. One of these will be connected to your LAN (Intranet), so there will be no restrictions on traffic between PBXpress and an IP phone. The other interface will be assigned a public IP address and connected to WAN (Internet). Thus, once again, there will be no restrictions on traffic between PBXpress and the remote gateway or IP phone.

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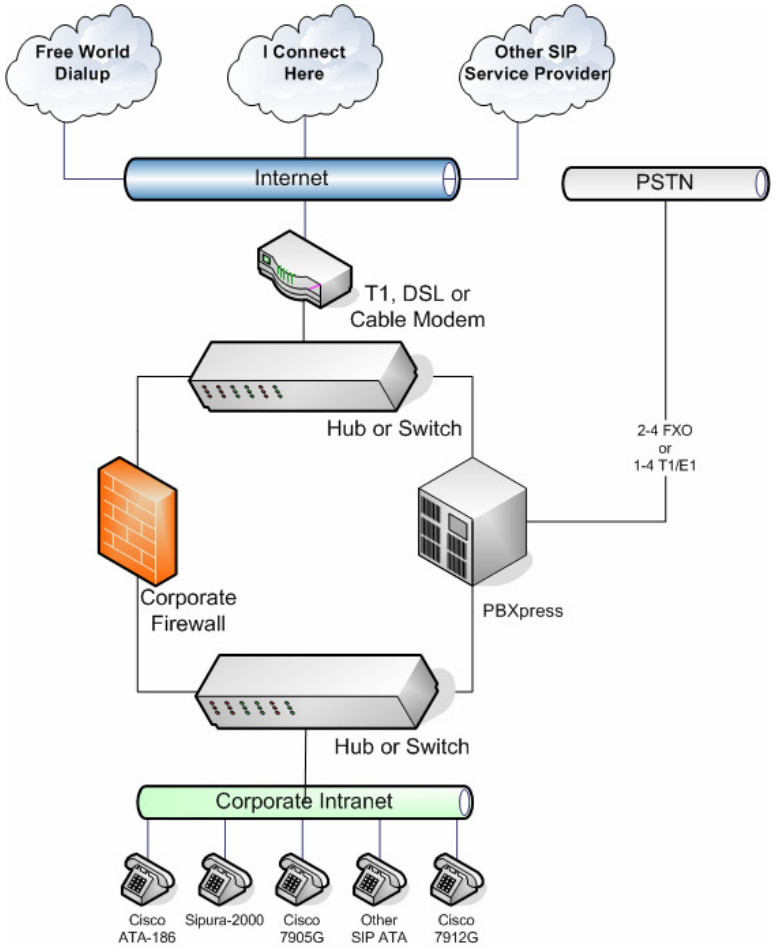

So when a call is made, PBXpress will maintain two separate calls:

- Between an IP phone on your network and PBXpress
- Between PBXpress and the remote party

and bridge them together.

The main concern when migrating to VoIP technology is reliability. Naturally, it is nice to save a few pennies on each call, but what if you could not make an important call because your Internet line was down, and lost a deal for USD 100,000? PBXpress allows you to start using VoIP without losing reliability. PBXpress will be connected to your existing PSTN lines (or some of them). If, for some reason, it is not possible to establish an outgoing call via IP, it will use one of the available PSTN lines. And, of course, it means savings when your Internet connection is up (which is normally most of the time). Anyway, that "missed important call" seems to be a thing of the past. PBXpress can take incoming calls and forward them anywhere via your IP phone: your office, home, or your notebook with a WiFi connection.

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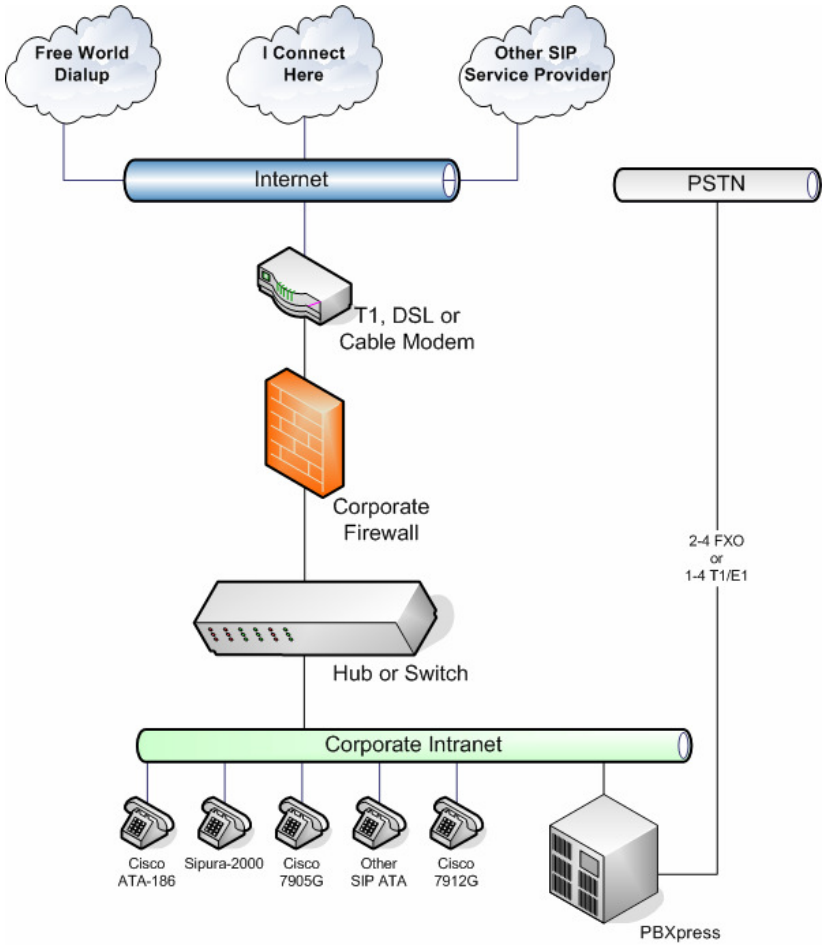
Internal Network Only

The fact that some box has access to both internal and external networks might raise some concern on the part of your security administrator. And even if the PBXpress box is running Unix, with a secure configuration that is quite difficult to break into, it is better to be safe than sorry.



In this case, PBXpress will be installed with a LAN connection only (of course, you have to make sure that your firewall allows VoIP communication between it and external VoIP carriers). Everything which was said in the previous section remains true: you can still use PSTN lines if your Internet service is down. With this method, you can still make outgoing calls to VoIP carriers, and even receive incoming calls from them.

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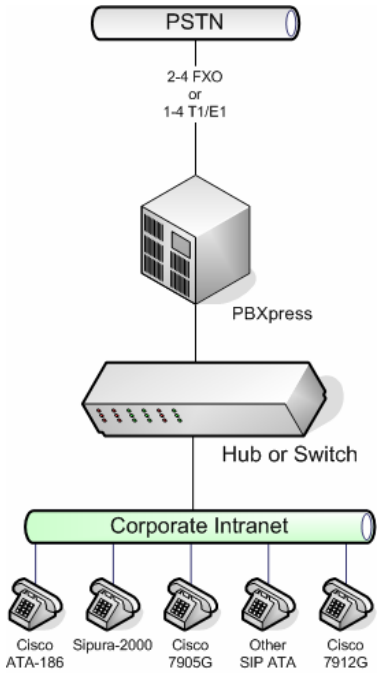
PBX Replacement – Without Outgoing VoIP

Even if you are not ready to start saving on outgoing VoIP calls at this moment, you still might find PBXpress extremely useful. If you are looking for PBX for a new office, why not use PBXpress instead of traditional PBX? Perhaps you do not have a high-speed Internet connection in your office at the moment; but shouldn't you prepare for tomorrow, when you will have one?



You get a solution which is:

- Cost-efficient
- Extensible
- Equipped with call recording capabilities
- VoIP-ready – after you get your Internet connection, you can send outgoing calls via IP with just a few mouse clicks, reducing your costs substantially

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Hardware Setup

Unpack PBXpress from the box and follow the further instructions in this section.

Networking Setup

Consult the *Usage* section above and decide on the deployment scheme for PBXpress.

Network Bridging

1. Plug the USB Ethernet dongle into the first available USB port using the USB cable, as shown below.

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2. Plug the WAN cable into the USB dongle, as shown below.

3. Plug the WAN Ethernet cable into a USB Ethernet dongle.
4. Plug the LAN Ethernet cable into an on-board Ethernet socket.

5. Install phone connectors according to the instructions in the next section.

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Internal Network Only

1. Plug the LAN Ethernet cable into an on-board Ethernet socket.

2. Install phone connectors according to the instructions in the next section.

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Phone Lines

PBXpress F400, 4 FXO

Use all available ports.

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Power Cable

Connect the power cable as shown in the picture below.

Sudden system shutdown due to a power outage can potentially cause problems on the PBXpress file system and prevent PBXpress from functioning normally. It is highly recommended to connect PBXpress to UPS (uninterruptible power supply).

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Check that PBXpress Has Obtained an IP Address

Upon successful connection to the DHCP server on your LAN, PBXpress will obtain an IP address, and this address will be displayed on the LCD.

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System Concepts

PBXpress consists of several components:

- Administrator web interface – what you use to manage the system
- IP PBX
- Call recording engine and call records
- Voicemail web interface

Administrator Interface

Easy to use, and accessible from any platform where a modern Internet browser is available, this interface allows you to:

- Change the system's global parameters (e.g. public IP address),
- Auto-detect and provision individual IP phones,
- Configure connections to external carriers you terminate calls to,
- Browse the call log history, listen to recorded conversations,
- Manage backup of call records on CD-ROM or DVD-ROM.

There are two modes for the administrator interface: express and advanced.

- In express mode, the web interface contains only the basic configuration screens needed to start using the service. Express mode is used by default.
- In advanced mode, you have access to all features and configuration options.

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IP PBX

PBXpress includes a SIP server with many features. SIP stands for Session Initiation Protocol; it is a protocol used by different elements of your VoIP network to communicate with each other. The SIP server provides several functions:

List of Extensions and Extension Groups

You may define a list of extensions (short numbers for dialing within your office environment) and assign a "real" phone number for every extension, ""so it can be reached from the outside world.

If your company has multiple "sites", departments, or other independently managed sets of extensions, each can be represented in PBXpress as an "admin group". This even allows you to provide "outsourcing" of IP PBX features, e.g. you can manage a set of phone extensions for your partners or resellers in your own PBXpress.

List of IP Phone Registrations

When Peter dials David's number, David's phone should start ringing and, when David answers, they should be connected. It is easy to do this with an analog PBX: some time ago it was entered into the PBX configuration that Peter's phone is connected to the cable via slot #25, while David's phone is attached via slot #46. So when Peter calls David, it is just a question of connecting these wires. If David decides to move to another office, or there is some other change, this requires PBX re-programming and, perhaps, re-installing the physical phone wires in your office.
With IP there is no physical connection between the user's IP phone and PBXpress. David can connect his phone in another office or in a meeting room. Or tomorrow, due to re-organization, you could completely re-design your office network. None of this matters so long as there is a working IP network between the users' IP phones and the SIP server. When David's IP phone goes online, it connects to the SIP server, informing it of the phone's current location (IP address). The SIP server remembers this information, and knows where a call should be routed to when someone calls David.

Number Translation

It is the SIP server's responsibility to know that if John dials 304, this is the abbreviated number for David, and so the call should go there. Thus

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the list of number extensions (abbreviated dialing) is maintained in only one place, inside PBXpress – there is no need to enter it into every individual IP phone.

Call Routing

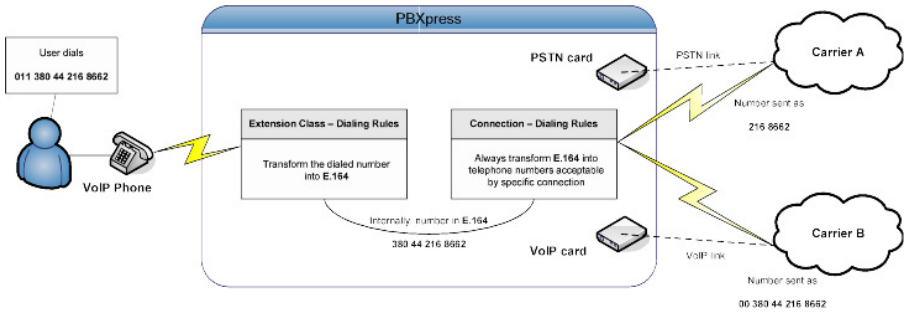
Someone tries to call Prague in the Czech Republic. Should the call go over IP to Packet8 or to iConnectHere; or is it better to just use your existing PSTN line? This will be decided by the SIP server, based on your connection settings.

List of Completed/Uncompleted Calls

Each call attempt is stored, so that later you can see who made the call, when the call was made, and to which destination.

Dialing Rules

In the modern telephony world, the same phone number may be dialed differently depending on the customer's habits or the carrier being used. For instance, when dialing a phone number within the Czech Republic, the number dialed will be 0016041234567. But the same number will be dialed as 16041234567 from the United States.
PBXpress dialing rules allow users to easily describe which numbering format should be used by specifying important parameters such as international dialing prefix. After this, PBXpress will be able to automatically convert a phone number into the correct format.



Connections

Connections define how the PBXpress system is connected to other telephone networks. There are two types of connections, depending on which protocol is used to carry the call: VoIP and PSTN. For instance, if PBXpress sends some calls via FXO ports to the local telco, and some other calls over IP to a VoIP carrier, there should be two connections in the configuration, with the PSTN and VoIP type, respectively.

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Accounts

Accounts are only applicable to VoIP connections. Since a VoIP carrier usually requires your VoIP box to authenticate itself in order to prevent service abuse, these credentials must be entered into the PBXpress configuration.

Outgoing caller ID

Caller identification includes two fields:

- CLI (ANI) number – the caller's phone number, e.g. 16045211234
- Display name (caller info) – usually a name and surname or a business name identifying the caller

When someone makes a call from an IP phone, information such as the phone number or the identification string "John Doe" is passed on along with other call information, and is visible to the called party. This works fine when making calls between phones within your organization, but carriers may apply certain restrictions. In this case, it may be necessary to alter the caller info before sending the call to the carrier. For instance, if you have a single phone line from a telco, all outgoing calls from IP phones made via this line should have the same CLI (ANI) number, i.e. the one which was assigned to you by the operator.

Dialing rules

Since different carriers may require phone numbers to be sent to them in a different format (e.g. a call to Seattle going via PSTN to Qwest should be dialed as 123-4567, but a call to the same number going via SuperPacket VoIP should be dialed as 206-123-4567), dialing rules must be used to ensure that PBXpress converts the number properly.

Out routing

This defines which calls should be dialed via a particular connection. For example, calls to the USA and Canada should be terminated to the local telco via a PSTN connection, while all international calls are to be terminated to a low-cost VoIP carrier. This is defined by specifying a set of phone prefixes for each connection. For instance, the phone prefix "420T" on connection "Nextra CZ" means that every phone number starting with 420 will go to this carrier. As a result, all calls to the Czech Republic will be terminated to this carrier.
Please note that prefixes are specified in the E.164 format – so the phone number begins with the country code, then the area code, and then the actual number. Thus the phone prefix for Washington D.C. is 1202T, and not 202T! (202T is, in fact, a prefix for Cairo, Egypt.)

Multiple connections and fail-over

It is possible that, in order to provide stable voice service, you will have several voice carriers available for the same destination. For example, you will specify prefix 420T for vendor SuperPacket VoIP (Czech Republic)

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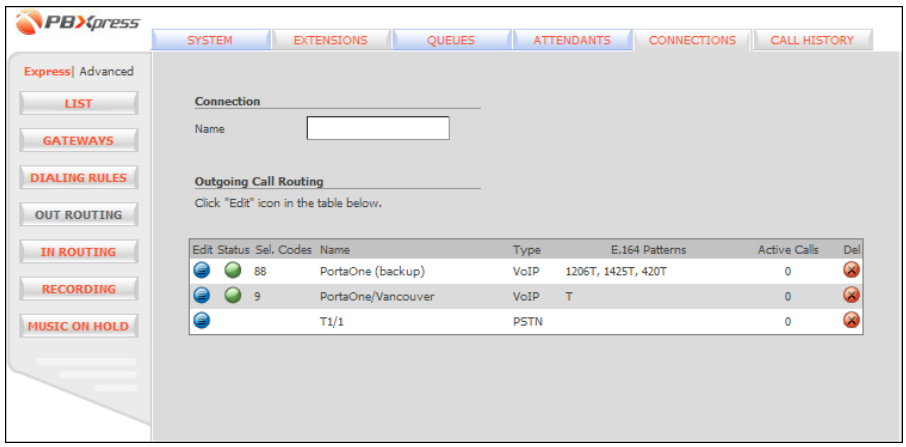
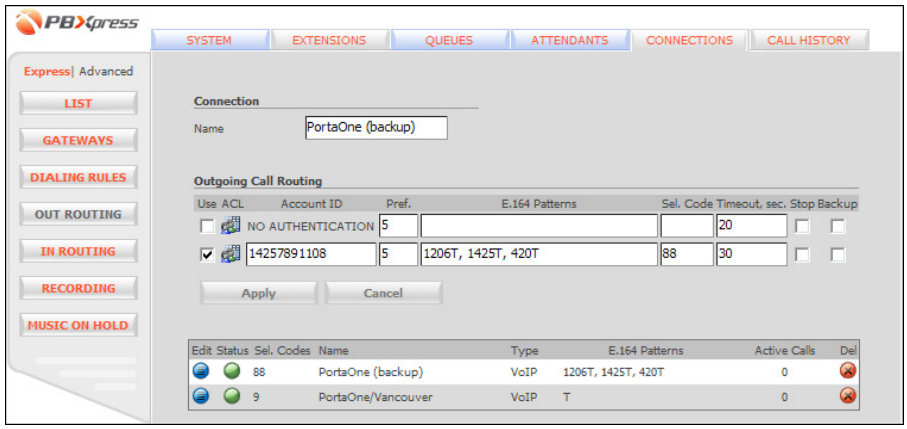
and prefix 4202T (Czech Republic, Prague) for vendor Call4Less, as shown below.



What happens if someone dials 42021234567? PBXpress will find a connection with the best matching prefix. In the example above, this would be Call4Less, since it has a 4-digit prefix match, compared to the 3-digit prefix match of SuperPacket VoIP. If several connections have exactly the same prefix, as shown in the next picture, the call will be routed to the connection with the higher preference.



Preferences are specified as positive numbers; thus, preference "5" for a connection means a more preferred route compared to another connection with preference "3".
It is also possible to maintain a certain connection as a "backup". This means that, if a call cannot be terminated via one of the connections according to your prefix specification, it will go via the backup one. This allows you to easily configure situations such as "if VoIP carrier fails, send calls via our office phone", without having to enter multiple prefixes in the telco connection configuration.

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In routing

In routing defines which IP phone (or phones) incoming calls will be connected to. Thus an incoming call to your office line #1 will ring on all the phones of people in the sales department, while office line #2 will be forwarded to Joe's personal phone.

Music on Hold

When one party places the other on hold during a call, you can configure PBXpress to play music for the party on hold. You can upload your own music files to provide some entertainment for the waiting caller, instead of just silence.

Queues

A queue is a logical group of extensions (IP phones) for the purpose of automatic call distribution (ACD). For example, if there are five persons working in the sales department and each one of them has an IP phone, it is highly desirable to transparently route incoming calls to any available phone(s) in sales.
In addition to specifying which phones are included in a queue, you should also specify in what manner PBXpress should ring the available extensions. The following possibilities exist:

- Simultaneous ringing on all the phones belonging to the queue (**ringall**). The first phone that answers will be connected.
- Ring one extension at a time. If that extension does not answer within a specified time interval, ring the next extension. In this case, the order in which extensions will be dialed may be controlled as follows:
    - Extensions are tried in the order in which they are defined (**roundrobbin**). For example, if the queue is defined as consisting of extensions 101, 102 and 103, PBXpress will first ring extension 101, then 102, then 103.
    - The list of extensions is sorted by the time since an extension last received a call, in ascending order (**leastrecent**). Thus, the extension which received a call the longest time ago will be on top of the list.
    - The list of extensions is sorted by the number of calls previously received by each extension, in ascending order (**fewestcalls**). Thus, an extension which received the least number of calls will be on top of the list.
    - The list of extensions is randomly shuffled for every new incoming call (**random**), so that the order will be different each time, and cannot be foreseen.

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Attendants

The attendant feature allows users to easily design and customize the IVR menus (e.g. "Press 1 for sales, press 2 for technical support") which will be launched when a call is made to a certain phone number.

Extensions

An extension is an abbreviated number which can be dialed to reach:

- A specific phone number provisioned on an IP phone
- A queue
- An attendant

PBXpress ensures the uniqueness of extensions. Thus, if extension 121 has been created for use on an IP phone, you cannot assign this number to any queue or attendant. Similarly, if the Sales queue has been assigned extension 201, then you cannot assign 201 to any individual phone.
For improved management of a large number of IP phone extensions, you may define extension classes, each of which describes a set of generic parameters (e.g. preferred dialing format, or whether long-distance calls are allowed). Next, a class may be applied to multiple extensions, automatically altering their dialing parameters.

Call Barring

The administrator may limit a certain extension's ability to make outgoing calls to certain destinations, e.g. allow local and toll-free calls, but disable long-distance and international calls.

Follow-me

The follow-me functionality allows callers to transparently forward a call to another destination if the called party's IP phone is not available. For instance, you may choose to forward calls from your SIP phone to your mobile phone. In this case, you will never miss an incoming call if you are not at your desk.
PBXpress' flexible follow-me allows every extension owner to provision his own set of multiple follow-me numbers, with a validity period assigned to each phone number (e.g. to prevent calls at night to your home phone number).

DISA

DISA (Direct Inward System Access) allows someone from outside the IP PBX to obtain an "internal" system dialtone and place calls as if from one of the phones connected to the switch. A user calls a number that

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connects to the DISA application, provides his extension number and password. If the password information is correct and the extension is allowed to use DISA service, the user receives a system dialtone on which a call may be placed.
You may allocate a dedicated external phone number solely for DISA purposes, so every incoming call to that number immediately connects to DISA application. Or, alternatively, if you have no spare incoming numbers, you may simply add an "unpublished" option to your main IVR (so you call to your company's main menu and then press 2 (technical support sub-menu) and then 8 for DISA service).
You may also use DISA from an IP phone by dialing the DISA access number. This is helpful when you are not at your desk, but at a different extension, which is not authorized to make a certain category of calls (e.g. international calls).

Call Parking

Call parking allows the operator to put a conversation on hold via the operator console. Later the call can be resumed from a different IP phone.

Call Pickup

Call pickup allows a user to answer to the call ringing on some other extension by dialing a pickup code. Two modes are supported for pickup:

- Group pickup, when the user dials just a pickup code (e.g. *8). In this case he may answer a call ringing on some extension without even knowing it's number.
- Directed pickup, when the user dials the pickup code followed by a specific extension number (e.g. *8103) – this would answer a specific call ringing on that extension (103 in this example).

Attendant Console

This allows an operator or secretary to easily manage all traffic in the IP PBX environment – see individual extensions and call status, answer calls, transfer calls, and so on.

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XML-RPC

The XML-RPC interface allows easy integration of external applications (such as customized operator console or CRM) with PBXpress. This application can obtain information such as a list of extensions or the status of current calls in real time and display it to an end user.

Call Recording

Every call which passes through PBXpress is recorded as a sound file on the internal hard drive. After this, you can browse call records and listen to conversations online.
However, even if the storage capacity of PBXpress is large, it is still limited. So the solution is to periodically move old records onto external media, such as CD-ROM or DVD-ROM, thus freeing up space for recording new calls. Each backup set is assigned its own unique ID. Thus you can still listen to a call which was made several months ago, and is no longer available on PBXpress' internal storage. You will be prompted to insert the CD or DVD with the required ID into the drive, and PBXpress will retrieve the necessary file automatically.

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Voicemail

When a certain extension is not available or not answering, the caller may be given the option of leaving a message, instead of just dropping the call. After that, the called party will be informed that there is a new message waiting, which he or she can retrieve and then take appropriate action.

Message waiting indicator (MWI)

The user will see directly on his/her phone that there is a new message. The actual representation depends on the phone model; it may be an envelope icon on the LCD, a special dial-tone, or something else.

Email notification

When a new voice message arrives, either a short notification or a complete voice message (a .WAV sound file in an email attachment) can be forwarded to the user's external email (e.g. on your company's email server).

Retrieving voice messages

The user can listen to his/her voice messages by:

- Dialing a voicemail access number from his/her IP phone.
- Logging into the voicemail web interface from his/her PC.

Queues and Special Settings (follow-me, voicemail)

What happens when an extension involved in one or more queues has the follow-me service enabled? In the case of queues, follow-me and voicemail settings are ignored, since they can confuse queue processing (especially in case of ringall queues). Also, if extension 123 is currently engaged in a phone conversation, and a call comes in to the Sales **ringall** queue, which includes this extension, 123 will not be notified.
If a person is not available on an IP phone, but still wishes to receive incoming queue calls, there is the possibility of specifying a "queue forwarding number" in the **Queues** configuration section for this extension. Please note that when this parameter is set up for a certain extension, all incoming calls from queues will be automatically forwarded to the specified phone number, instead of to the IP phone.
Of course, an extension's voicemail settings are ignored during queue processing (otherwise the call will always be terminated to the first extension with voicemail activated).

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Cluster Environment

You may connect several PBXpress instances into a dynamic cluster. This is very useful in case of a multi-branch enterprise with several offices. Each location should have its own PBX to perform local administration tasks and yet, at the same time, the company's system should work as a single entity (e.g. when a new person joins the company at location A and the local administrator provisions his phone - there should be no need for administrators at locations B, C and D to provision the same information again in their systems).
PBXpress cluster is comprised of nodes, where each node is a PBXpress system with a unique name. The node can play two roles:

- Boot node serves as a repository of data on the current cluster configuration and provides this information to other nodes.
- Slave node periodically connects to its boot node to submit information about the local configuration (extensions, connections, etc.) and to obtain information regarding other nodes.

Thus, when you create extension 101 on node A, this information will shortly become available at all other nodes in the cluster - and if someone on node B now dials "101", this call will be forwarded to node A and connected to the corresponding IP phone there. PBXpress cluster can detect cluster configuration conflicts (e.g. extension 102 is created on node A, but the extension with this ID already exists on node B.
Node A can serve as a boot node and at the same time be a slave node to some remote node. Typical cluster configurations are:

- Dual – the simplest possible cluster configuration, one boot and one slave node.
- Star – one central boot node and multiple slave nodes connected to it.
- Tree – Several "branch" nodes connected to the central boot node ("root") at the same time serving as boot nodes for other nodes ("leaves").

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PSTN Interface Cards

If a call cannot be delivered over the Internet, PBXpress can use your existing PSTN lines to complete it. Since a SIP server is an IP-based solution, additional hardware modules must be installed into PBXpress so as to interface with the PSTN network. The type of these modules (ports) depends on which phone lines you currently have:

- FXO (Foreign Office Exchange) – this is a port where you can connect your normal telephone line (the one your analog phone was plugged into). So if your office used to have several phone lines from a local telco operator, you can connect these to PBXpress, and PBXpress will utilize them.
- BRI - Basic Rate Interface. ISDN BRI consists of two 64 kb/s B channels and one 16 kb/s D channel for a total of 144 kb/s. This basic service is intended to meet the needs of most individual users. BRI is used predominantly in Europe.
- T1/E1 – this is a connection where a single set of physical wires can carry many simultaneous calls. Normally, this is available for larger companies and bigger phone switches. PBXpress support both ISDN PRI and CAS signaling over a T1/E1 connection.

PBXpress is supplied in several configurations, with different PSTN interface cards to accommodate your needs. The advantage ISDN voice interfaces offer over other analog voice interface types is that ISDN signaling provides end-to-end call information, with details of the called and calling numbers, numbering types, call progress status, and call clearing cause codes. Also, the audio is sent as a digital pulse code modulation (PCM) stream so problems due to audio levels and attenuation are reduced.

Faxes

When the incoming call is connected to voicemail or auto-attendant – caller can initiate a fax transmission at any time, PBXpress will automatically switch to fax mode and then the received fax will be stored as a new message with PDF attachment in the mailbox. In the auto-attendant menu there is a special event "Fax" which is triggered upon fax transmission. Action for this event will define a voice mailbox, where the received fax document will be stored.

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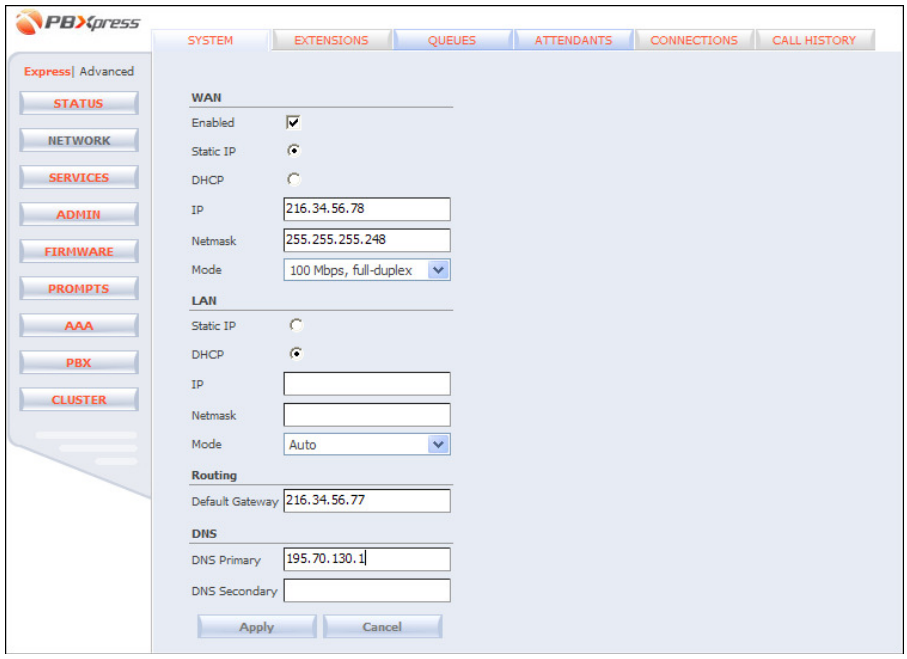
System Configuration

Using the Admin Web Interface

Navigating the Admin Web Interface

All PBXpress admin screens look similar. At the top there is a list of tabs (e.g. **System**, **Extensions**, etc.), which define different parts of the system configuration. To switch between tabs, simply click on the name of the tab required.
When a certain tab is selected, you will see a list of different configuration screens in the left column (such as **Status**, **Network**, **Services** - see the picture below). To switch between configuration screens, just click on the button with its name.



If you have changed something on a screen, be sure to press **Apply** before going to another configuration screen.

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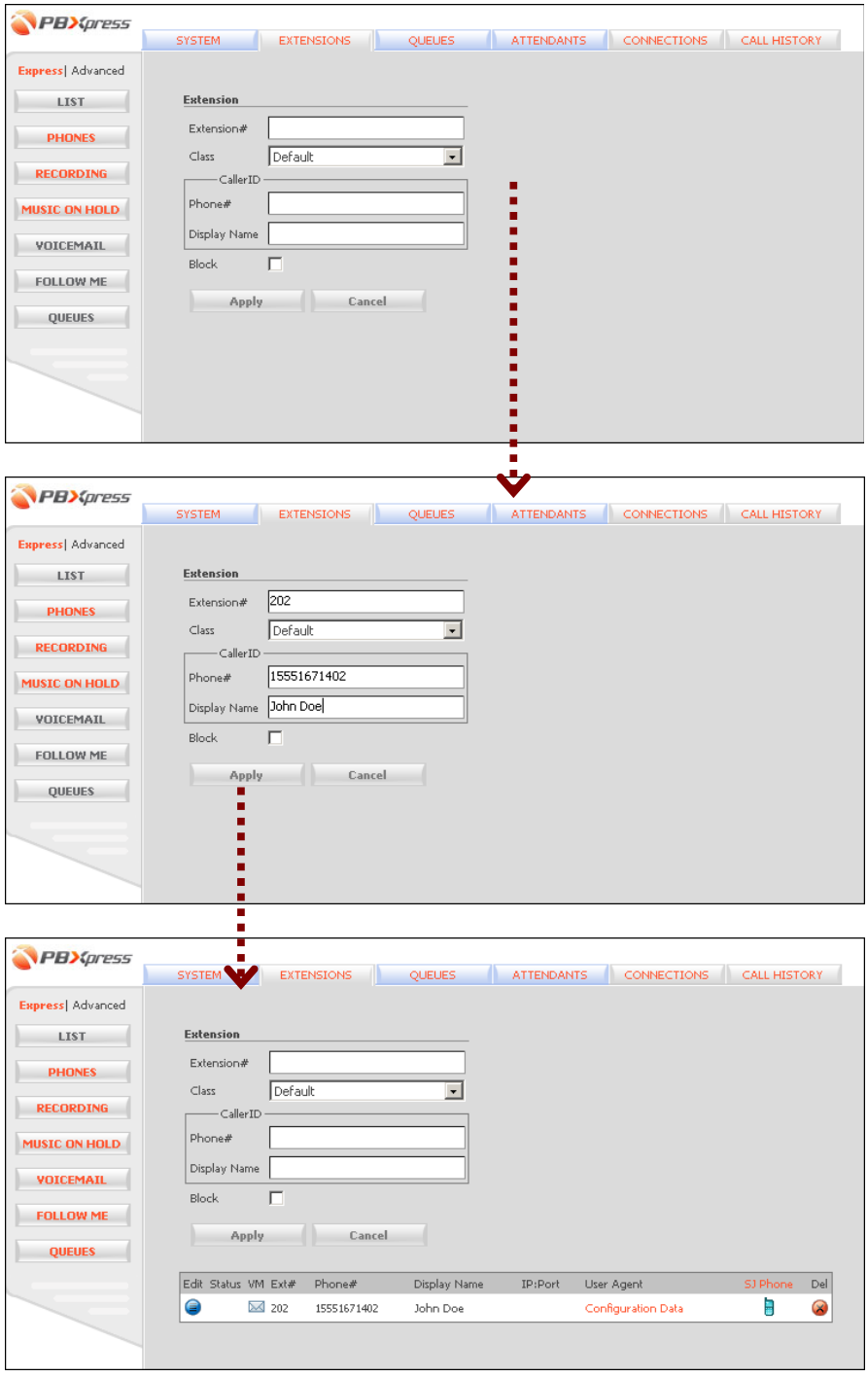
Adding New Records

Usually, when you open a form it is in the mode for creating a new record. Enter the required values in the available fields and then press **Apply** to save the changes.



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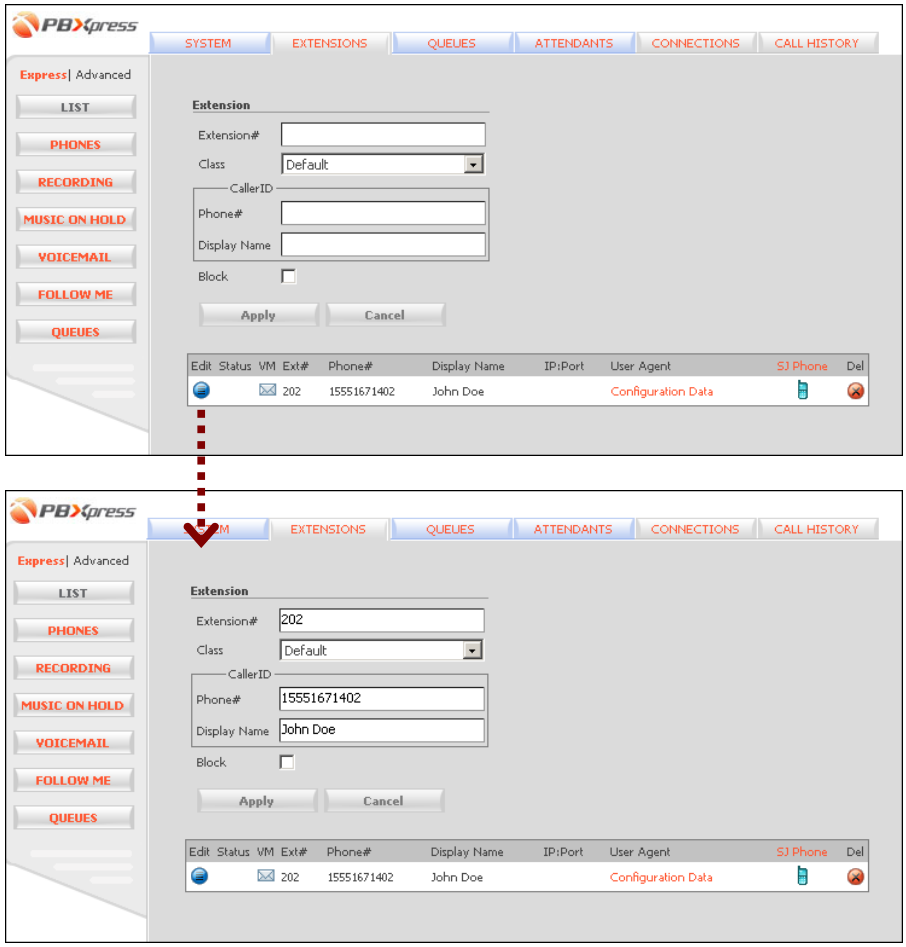
Editing the Existing Record

As already mentioned, when you enter a form it is in "add new" mode. If there are some existing records, you will see them in the table at the bottom of the screen. To edit a certain record, click on the Edit icon beside it.



Press **Apply** to save your changes.

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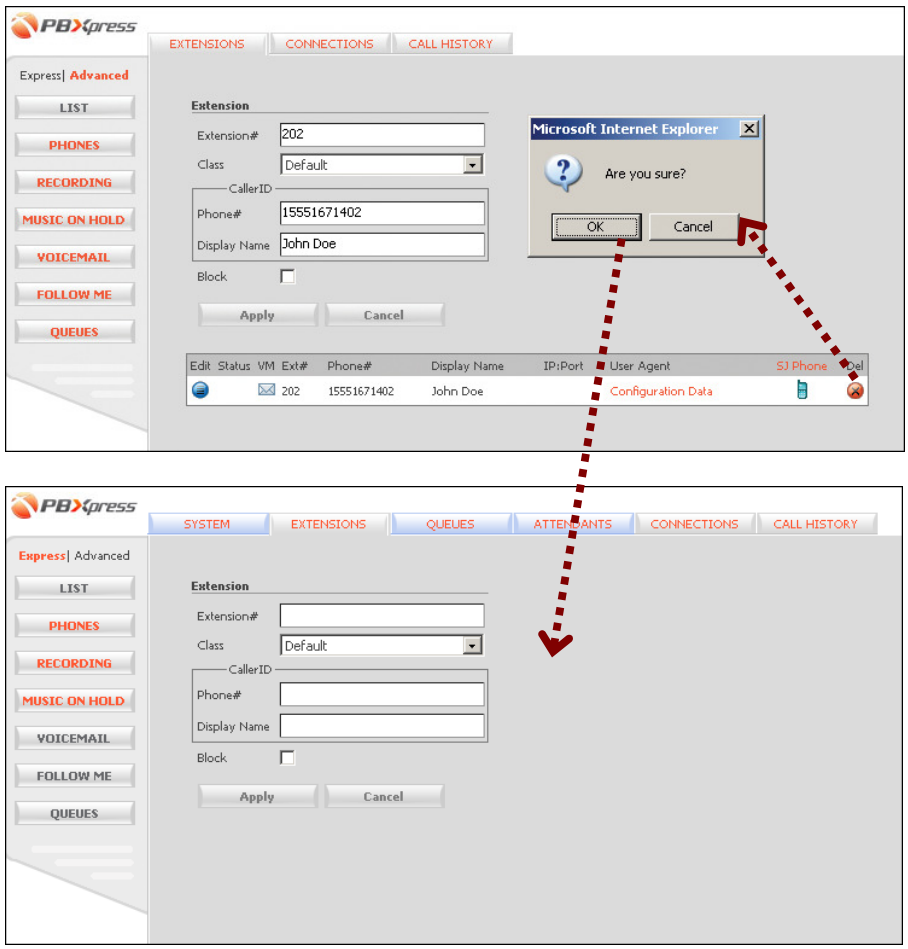
Deleting an Existing Record

In the list of available records, the Delete icon appears in the rightmost column. This icon allows you to delete a specific record. If the record cannot be deleted because it is referenced by another configuration element, the Delete icon will be absent.



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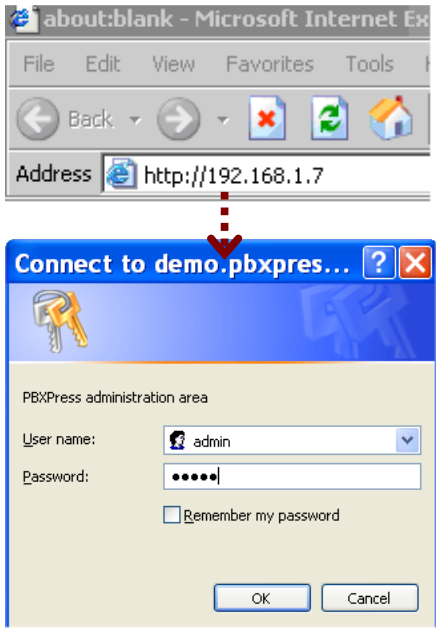
Initial Configuration

Connect the internal network PBXpress interface to your LAN, making sure that there is a DHCP server available. Turn on PBXpress, and wait until it finishes booting up. The PBXpress IP address will be visible on the LCD display, as shown below:

Now, open a web browser on the PC connected to the same LAN and type in **http://<IP>**, where <IP> is the IP address you saw on the LCD display (see more about this in the *Check that PBXpress Has Obtained an IP Address* section in the *Hardware Setup* chapter). The admin interface is password-protected. The default username is "`admin`", and the password is "`admin`" as well.



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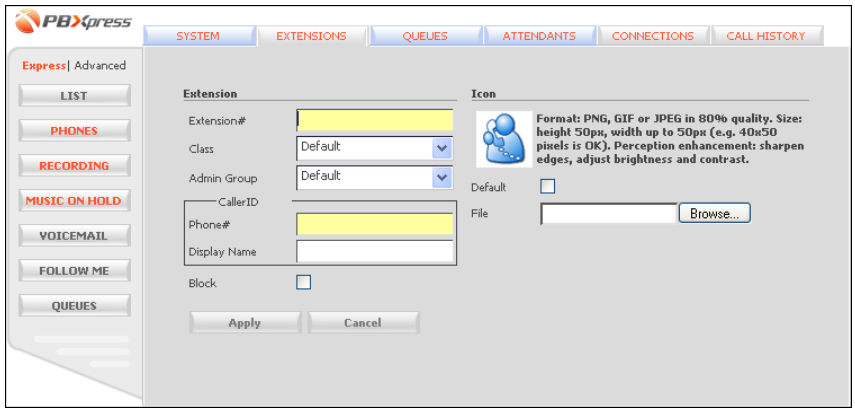
After this you will see the PBXpress admin main menu.



The following section gives you a detailed description of all the options available in advanced mode. To switch to this mode, click on the **Advanced** button, so that you see a PBXpress screen similar to the one below. Consult the PBXpress Quick Start Guide for a description of express mode.

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General Configuration

Status

This is a read-only page which provides information about the current status of the network and other system parameters.

If there is a smart model of APC UPS connected to PBXpress via a USB port, you can see its current status in the bottom group.

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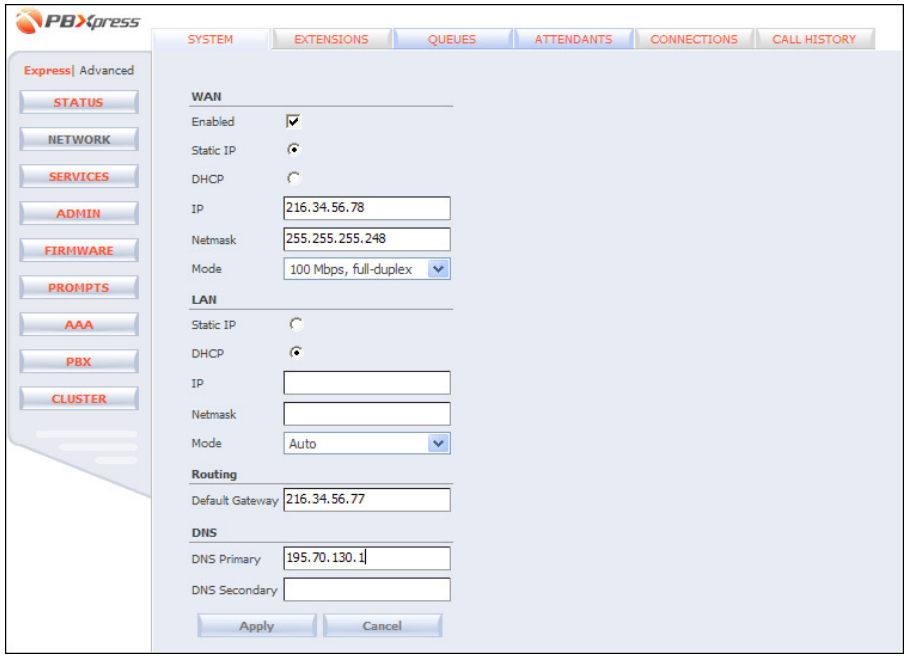
Network Configuration

Press the **Network** button on the left to access the configuration menu for network interfaces (WAN and LAN).



If you are using a WAN (Internet) interface, then the configuration values for the **IP**, **Net mask**, **Default Gateway** and **DNS** servers should be provided by your ISP or network administrator.
For a LAN interface, PBXpress will usually obtain an IP using your existing DHCP server, so that the default settings are fine.

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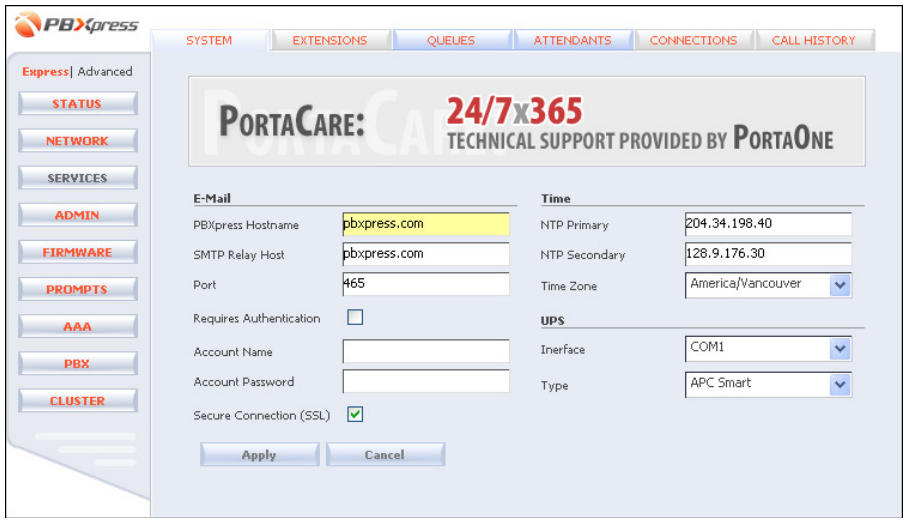
Service Configuration



Email

PBXpress can deliver email messages to external email servers (e.g. administrative notifications or voice messages). In order to do so, the email system must be properly configured, otherwise it will not be able to deliver email messages to the external mail server.

Field	Description
PBXpress Hostname	When sending notification emails, PBXpress must include information about itself in the email. Please make sure the name you assign to it (e.g. pbx.domain.com) is properly registered in the DNS.
SMTP Relay Host	For your users to receive notifications of new voice messages, PBXpress must be able to deliver notification emails to external mail servers (e.g. gmail.com). The best way to do this is to forward these emails through your company's or your ISP's mail server. Enter the name of this server here.
Port	SMTP port on the relay host; 25 by default.
Requires Authentication	Some ISPs may require user authentication before sending an email message, to prevent spam and email abuse.
Account Name	The username to be used for SMTP authentication (should be provided by your ISP).
Account Password	Password for SMTP authentication (should be provided by your ISP).
Secure Connection (SSL)	Activate this checkbox if your ISP provides a secure, encrypted way to communicate with its mail server.

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Time

In order to maintain the correct call connect/disconnect times, the PBXpress server must synchronize its internal clock with other servers on the Internet. The default settings are usually suitable for most users.

Field	Description
NTP Primary	Address of the primary time synchronization server.
NTP Secondary	Address of the backup time synchronization server.

UPS

This defines the connection interface to the UPS.

Field	Description
Interface	The physical interface on the PBXpress server which the UPS is connected to (USB, COM1, COM2, etc.).
Type	UPS model used – APC Smart UPS or APC Back UPS.

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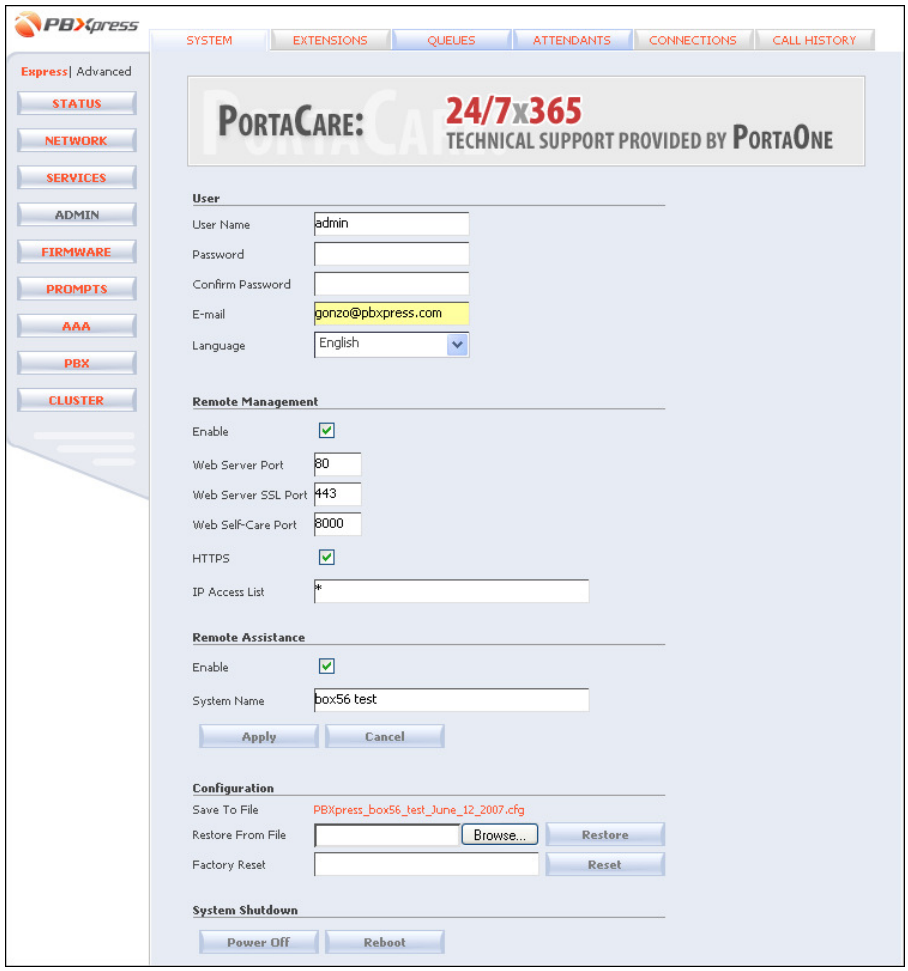
Administrator Access Options

This section controls access to the PBXpress administrator interface.



User section

Here you can change the user name and password used to access the admin interface. The email address you enter here will be used for:

- Sending alerts if the power is out and PBXpress is running on UPS power.
- Sending notifications about new available firmware upgrades.
- Password recovery.

Here you can also change the language of the web interface.

NOTE: Make sure you remember the new username and password after you change them.

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Remote Management section

Field	Description
Enable	If you would like to access the PBXpress admin interface both from your internal network (LAN) and from other computers on the Internet (e.g. from your home DSL line), enable remote management (it is turned off by default, so the admin interface is only accessible from the internal network).
Web Server Port	The network port for unencrypted (HTTP) administrator access.
Web Server SSL Port	The network port for secure (HTTPS) administrator access.
Web Self-care Port	The network port for self-care extension web access.
HTTPS	If activated, switches the web interface (both administrator and self-care) to secure mode (with SSL). All login attempts to the **Web Server Port** will be automatically redirected to the **Web Server SSL Port**, and for the self-care interface SSL version will be available on the port, following the **Web Self-Care Port** (e.g. if **Web Self-Care Port** is 8000, then SSL self-care will be running on port 8001)
IP Access List	It is highly recommended to list the IP addresses of all hosts allowed to access the admin interface in the **IP Access List** field. Then, if an intruder (even the one who knows the username and password) tries to access the system from his/her computer, the attempt will be rejected.

Remote Assistance section

In some cases the PBXpress team may need to remotely login to your system to perform troubleshooting or maintenance. To enable such remote access, activate this option and enter a descriptive ID in the **System Name** field (e.g. your company name), so we can identify your PBXpress from among others. By default this option is turned off, so that no one can access your system.

Configuration section

Here you can:

- Save the current PBXpress configuration (click on the file name in the **Save To File** line). For convenient administration of a multi-site PBX installation, the file name includes the PBXpress system name and the current date.
- Restore the PBXpress configuration from a backup file.

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Factory reset

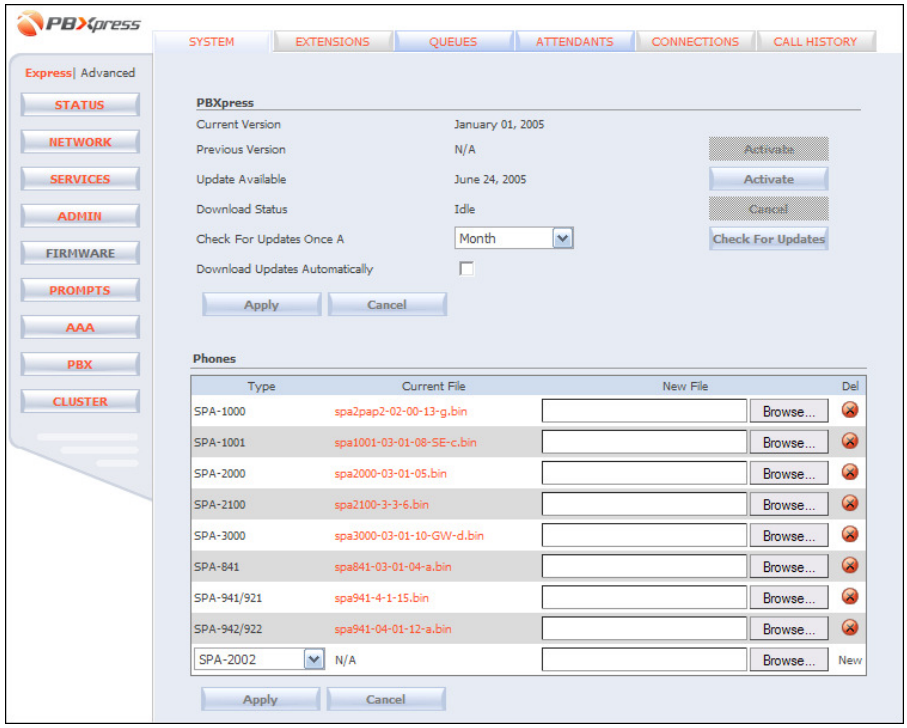
Enter *Factory Reset* in the **Confirm** field, and press the **Reset** button to clear the current PBXpress configuration and start system configuration from the very beginning.

System shutdown

You can remotely power off or reboot the PBXpress server.

Firmware

This form allows you to manage updates of the PBXpress software as well as firmware, installed on your IP phones.



This screen allows you to manage PBXpress firmware updates; see the *Performing a Firmware Update* chapter for more information.

Phones

You may designate a firmware image to be automatically provisioned on every supported IP phone model. PBXpress provides extra security for the phone firmware download – only phones currently registered to PBXpress, may download the firmware image.

NOTE: If you have phones provisioned with PBXpress version prior to 2.0, you need to reset the phone configuration (please, unregister the extension from the phone and re-assign the extension to the phone) to enable the auto firmware upgrade.

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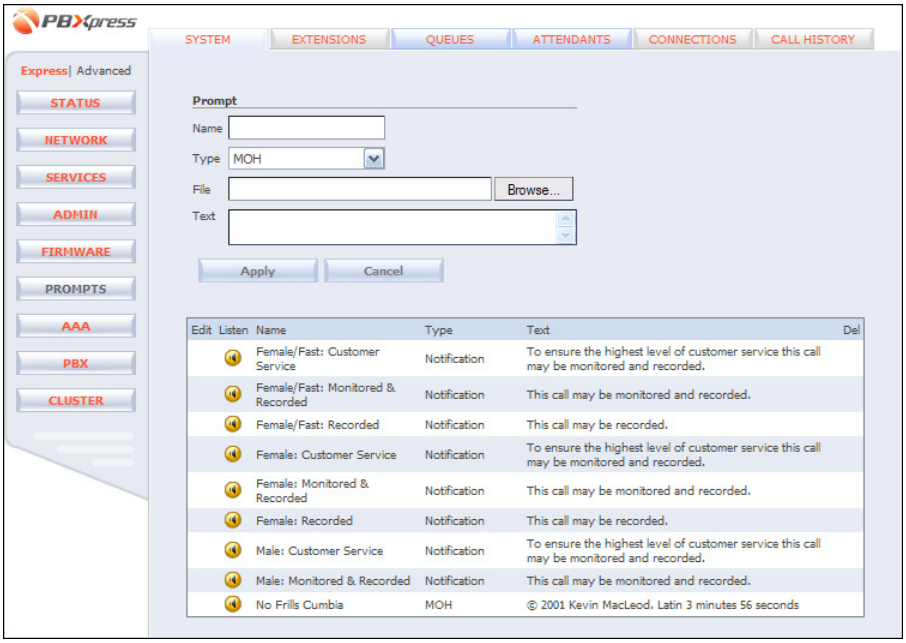
Prompts

Here you can manage the set of voice prompts to be played to a calling (or called) party. For instance, you can play an announcement stating that the call may be recorded. For easier management every voice prompt has a type assigned to it, which describes its intended use – e.g. music on hold or auto-attendant menu.



There are several pre-configured voice prompts supplied with the system, and you may also upload your own. When uploading a custom prompt, you can give it an abbreviated name and specify the path to the original sound file on your workstation. It is highly recommended that you type the full prompt text in the **Text** field, so that it is stored in the database.

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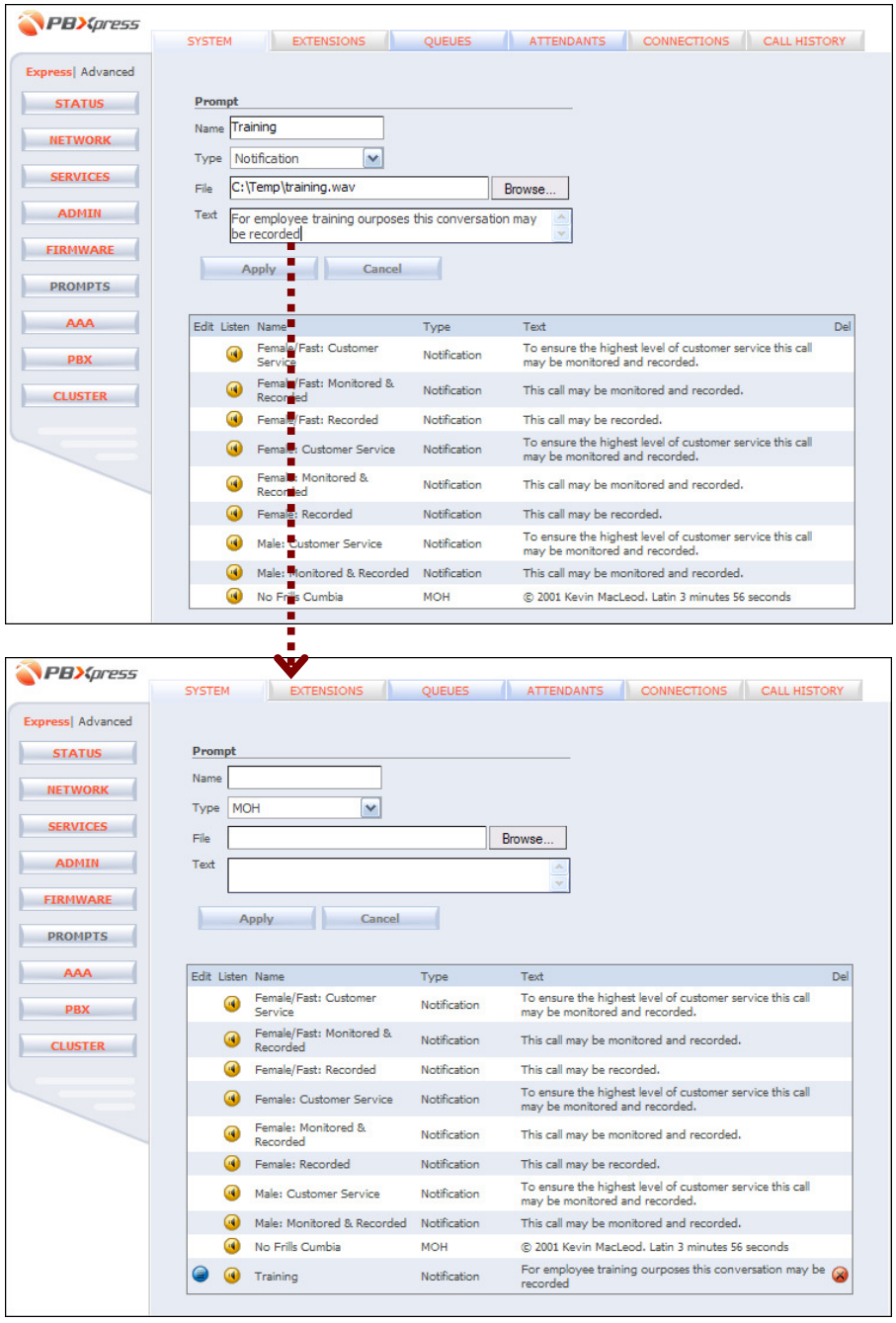

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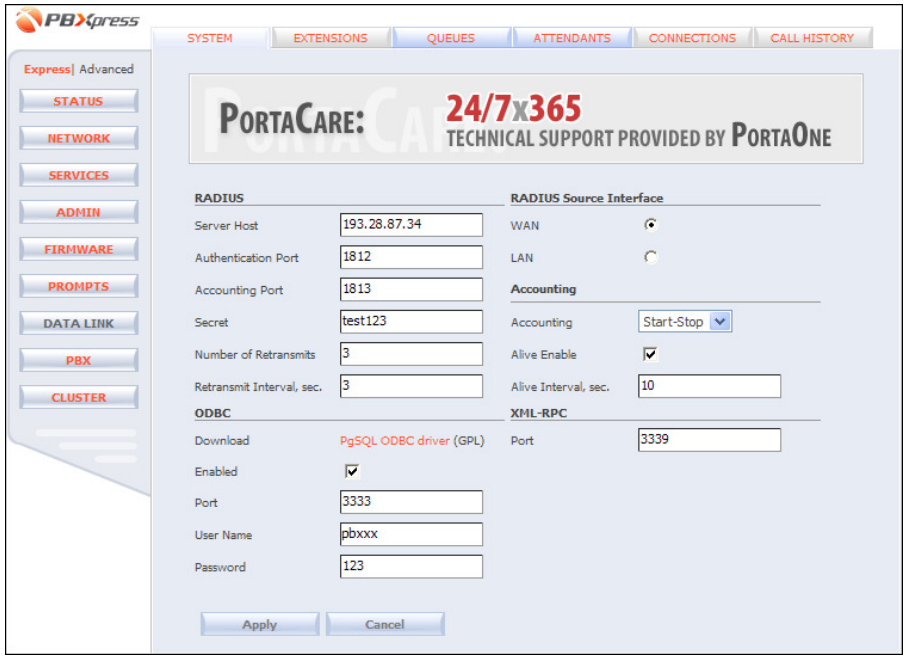
Data link

PBXpress may connect to an external RADIUS server for call authorization and accounting or to allow CDR data collection via ODBC.



Field	Description
***RADIUS** section*	
Server Host	Hostname of the RADIUS server. If you are using a hostname, make sure it is correctly registered in the DNS. Otherwise, use the IP address.
Authentication Port	The port on which the RADIUS server listens to authentication requests. 1812 is the default port according to the new RFC, while some older servers may use port 1645.
Accounting Port	The port on which the RADIUS server listens to accounting requests. 1813 is the default port according to the new RFC, while some older servers may use port 1646.
Secret	RADIUS secret key; make sure the value you enter here matches the one specified in the RADIUS server configuration.
Number of Retransmits	How many times PBXpress will try to resend each RADIUS request if no response is received from the RADIUS server.
Retransmit Interval, sec.	How long PBXpress will wait for a response from the RADIUS server before the next retransmit attempt.

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RADIUS Source Interface	Which network interface (**LAN** or **WAN**) should be used to send RADIUS requests.

Field	Description
Accounting *section*	
Accounting	Controls the way in which accounting packages are sent to the RADIUS server: • **None** – Disable call accounting. • **Start-Stop** – Send one accounting immediately after the call is connected (start accounting) and another one when the call is disconnected (stop accounting). • **Stop-Only** – Send one accounting record when the call is completed (disconnected).
Alive Enable	While the call is in progress, periodically send keep-alive RADIUS requests, informing the RADIUS server of the current call status.
Alive Interval, sec.	How often keep-alive packages should be sent.

NOTE: Call authentication and authorization are not currently supported.

Field	Description
ODBC *section*	
Download PostgreSQL ODBC driver for Windows	Click on the link to download the current ODBC driver for Windows, which will enable you to connect to the PBXpress database from your workstation.
Port	Which port the database should accept incoming connections on.
User Name	Login name for incoming database connections.
Password	Password for incoming database connections.

Field	Description
XML-RPC *section*	
Port	The TCP port on which PBXpress will accept incoming XML-RPC requests.

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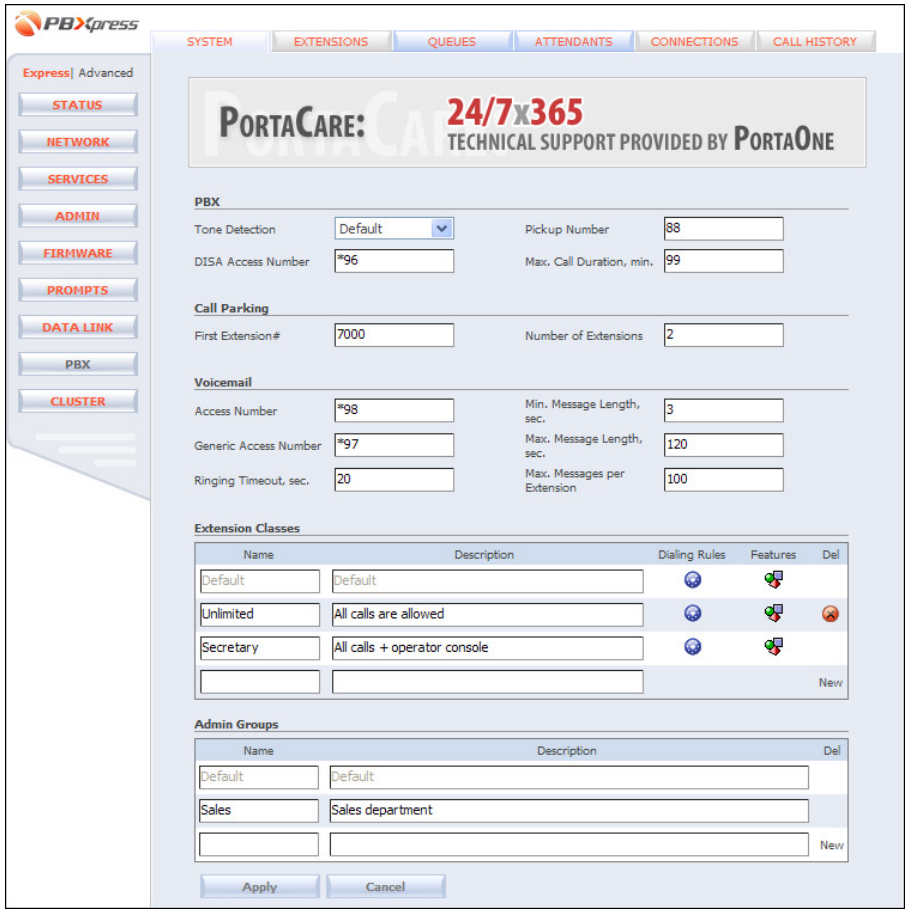
PBX

Here you may configure the parameters for such PBX features as the voicemail system.



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PBX

Field	Description
Tone Detection	Specifies how PBXpress recognizes signals from analog phone lines (can be used in some countries with non-standard signaling).
DISA Access Number	Direct Inward System Access number. When a user dials this number from his phone or is connected to this number from the auto-attendant menu – he will be prompted for his extension number and password and then can make an outgoing call.
Pickup number	A digit combination (prefix) to be dialed from an IP phone to pickup an incoming call to some other extension. See the *Call* section for more details.
Max. Call Duration, min	Allows to prevent excessively long calls.

Call parking

Field	Description
First Extension #	Every parked call is assigned a specific extension number for identification purposes. These extension numbers cannot overlap with extensions used for IP phones or auto-attendant queues. This parameter defines the extension number to be assigned to the first parked call. If there are several concurrently parked calls, they are assigned subsequent extensions numbers.
Number of Extensions	Maximum number of simultaneously parked calls.

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Voicemail

Field	Description
Access Number	Which number should be dialed from an IP phone in order to listen to the user's voice messages.
Generic Access Number	Which number should be dialed from an IP phone in order to connect to the generic voicemail IVR (where you will be prompted to enter your extension number and password). Typically, this is used when checking voice messages from somebody else's IP phone.
Ringing Timeout, sec	Default timeout parameter. If a call is not answered within this interval, the call is forwarded to follow-me number or voicemail. Please note that this parameter is only used for calls between extensions. For incoming calls, the timeout parameter is defined in the connection's **In routing** settings.
Min. Message Length, sec.	Too short messages (shorter than the specified duration) will not be allowed.
Max. Message Length, sec.	Limit the maximum length of the recorded message (to the specified duration).
Max. Messages per Extension	Allow no more than a certain number of messages to be stored in each mailbox.

Extension Classes table

NOTE: The pre-defined extension class Default cannot be deleted or changed.

Column	Description
Name	Name of the extension class.
Description	Description of the extension class.
Dialing Rules	Click on the icon to invoke the dialing rules wizard (see below).
Features	Click on the icon to define which call features to enable or disable for extensions of this class (see below).

NOTE: PBXpress can recognize whether a number is local, toll-free or long-distance/international only if dialing rules have been properly defined. If simple E.164 dialing is used, it is unable to distinguish between different call categories.

Admin Groups table

NOTE: The pre-defined Default group cannot be deleted or changed.

Column	Description
Name	Name of the admin group.
Description	Description of the admin group.

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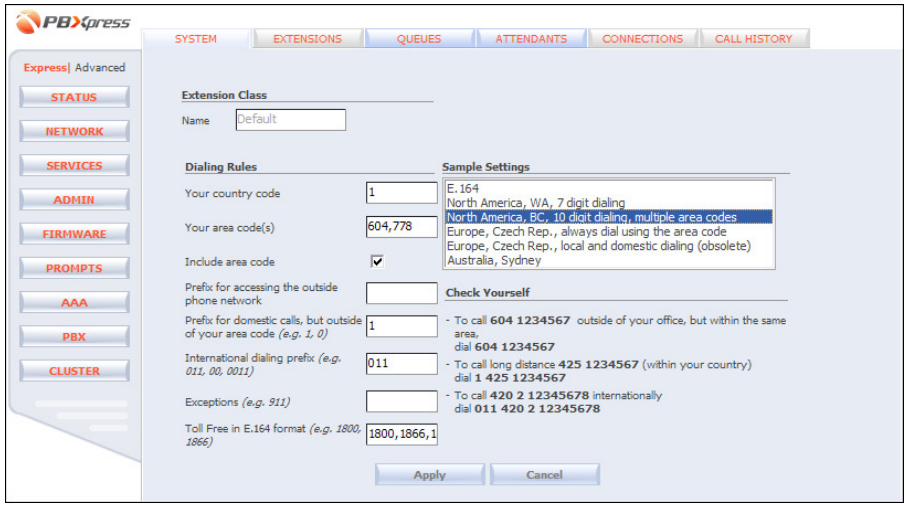
Dialing rules wizard

People around the world dial phone numbers differently. For instance, if Jan lives in Prague in the Czech Republic, and his home phone is 23456789, his friend John from Canada will dial his number as 01142023456789, and his business partner Hans from Germany will dial it as 0042023456789, while his girlfriend Zuzana in Prague will just dial 23456789. If IP telephony technology meant that each user had to be trained to dial phone numbers in a new way, this would overcomplicate the migration process. PBXpress allows companies anywhere in the world to use their preferred dialing format, so that nothing changes for the end user.

On the Dialing Rules configuration screen, you can specify parameters so that the system will know how to interpret dialed phone numbers.



Most of the parameters here are self-explanatory; if uncertain, you may load and learn the sample settings provided. Also note the **Check Yourself** area for verifying that the system has correctly "guessed" the way you dial phone numbers.

Field	Description
Your country code	The E.164 country prefix, e.g. 1 for USA & Canada, or 420 for the Czech Republic.
Your area code(s)	The code for the local area or city. If more than one area code is considered local (for instance, the area codes 604 and 778 in Vancouver, BC), enter all of them, separated by commas.
Include area code	If checked, this means you always dial the number with the area code, even when dialing within your "local" area. For instance, even when dialing a local number within the 604 area code, you must still dial 604 NNN-

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	NNNN.
Prefix for accessing outside phone network	Dial (if applicable) to get an outside line (e.g. in some office PBX systems, you must first dial 0 and then the actual number).
Prefix for domestic calls	The prefix to dial before the phone number when making domestic long-distance calls (e.g. 1 in the USA & Canada, 0 in many European countries).
International dialing prefix	The prefix to dial before the phone number when making international calls (e.g. 011 in the USA & Canada, 00 in many European countries).
Exceptions	A comma-separated list of special numbers. These numbers will not be handled according to dialing rules, but rather processed "as-is". For example, if you set up for the USA (country code 1) and the 206 area code, and a user dials 911 on his phone, the system should not convert this number to something like 1206911; rather, the number should stay 911. If you have multiple similar phone numbers with minor differences, you can use pattern symbol **T** when describing a phone number. It means that the phone number at this location can have any sequence of digits (including an empty sequence), e.g. 1555T will match 155512345, 15551 and 1555.
Toll Free	List of phone prefixes which are considered toll-free (e.g. 1800).

When **Include area code** is enabled, the **Exception** field is disabled, since in this case it is obsolete: PBXpress already knows that every phone number which does not start with an area code or a domestic/international dialing prefix is an exception.

NOTE: You must properly configure routing in the connections for exceptional numbers in order for users to actually reach their destination (e.g. emergency services).

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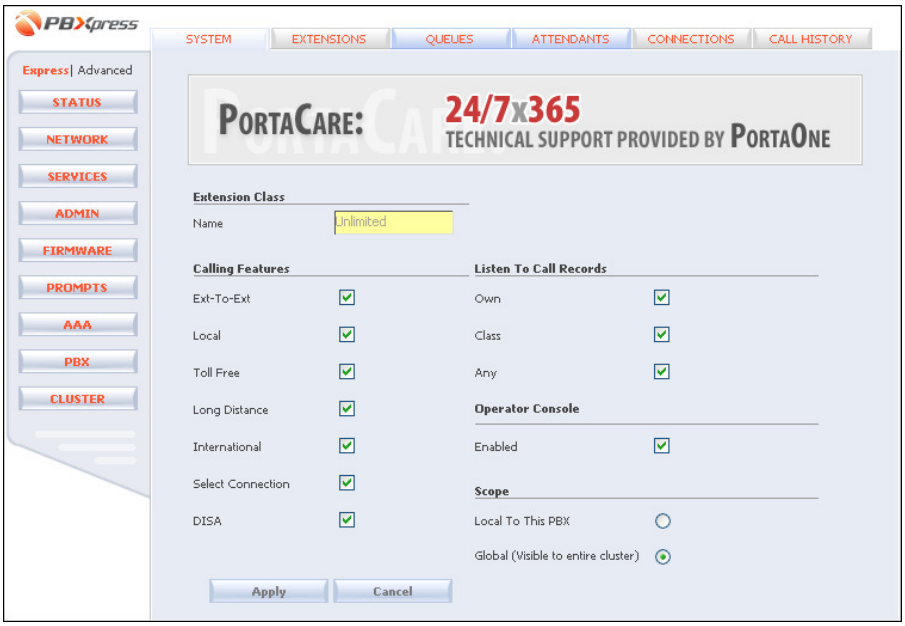
Extension class features



Calling Features

Field	Description
Ext-to-Ext	Enables calls to other extensions.
Local	Enables outgoing local calls.
Toll Free	Enables outgoing calls to toll-free numbers.
Long Distance	Enables outgoing domestic long-distance calls.
International	Enables outgoing calls to other countries.
Select Connection	Enables selection of a particular outgoing connection by dialing a connection prefix before the phone number.
DISA	Allows to use the direct inward system access feature.

Listen To Call Records

Field	Description
Own	Allows the extention owner to listen to recorded conversations.
Class	Allows the extension owner to listen to recorded conversations of any other extension in the same class.
Any	Allows the extension owner to listen to recorded conversations of any other extension.

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Operator Console

Field	Description
Enabled	Enables or disables access to the operator console for extensions within this class.

NOTE: Since the operator console enables the handling of all calls within a specific admin group, by default most IP PBX users should not be allowed to access this feature. Create a separate extension class ("secretary" or similar), then enable this feature in the extension class and assign it only to specific extensions.

Scope

Defines visibility of extensions of this class

Field	Description
Local To This PBX	Information about extensions of this class is not propagated to other nodes in the cluster.
Global (Visible to entire cluster)	Information about extensions of this class is distributed among all other nodes in the cluster.

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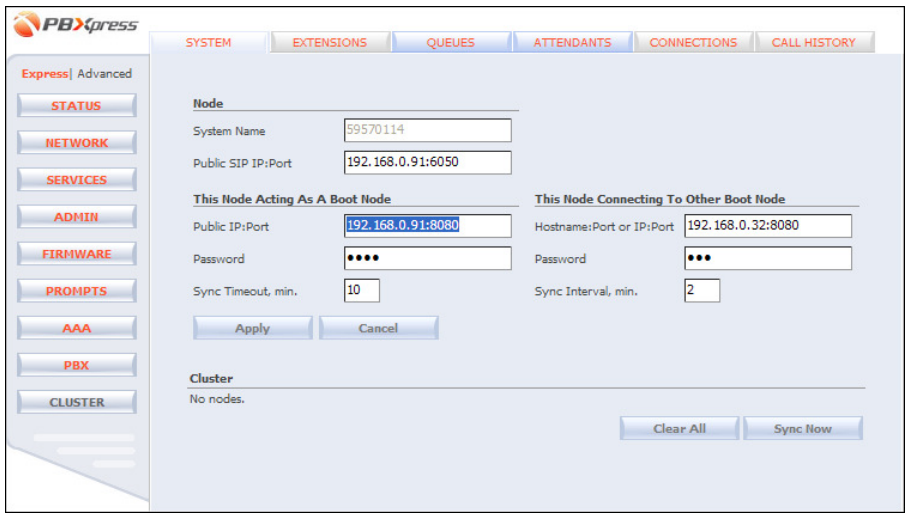
Cluster

You can configure the cluster environment by specifying whether this PBXpress serves as a boot node for other PBXes and what its own boot node is. For typical configurations (dual and star) you will usually fill in only half of the configuration form on every PBXpress.



Field	Description
System Name	Unique node identifier within the cluster (read-only – to edit go to **System / Admin**).
Public SIP IP:Port	Defines a network address to be used by other nodes in the cluster to send calls to this node's extensions. If the PBX is connected to WAN, you may leave this field empty, the correct value will be assigned automatically. If the PBXpress is only connected to LAN, you should set up the port forwarding on your router, so every incoming connection to NNN port will be transported to PBXpress port 5050. Then enter your router public IP and the NNN in this field.

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The Node Acting As a Boot Node

This allows you to set up this PBXpress as a boot node.

Field	Description
Public IP:Port	Network connection parameters for this node accept information requests from slave nodes
Password	To enhance security all communication between slave nodes and the boot node are protected by password.
Sync Timeout, min.	A slave node is considered to be off-line if it does not connect to the boot node within the specified time interval after the next scheduled re-sync time (re-sync interval is specified individually in every slave node configuration).

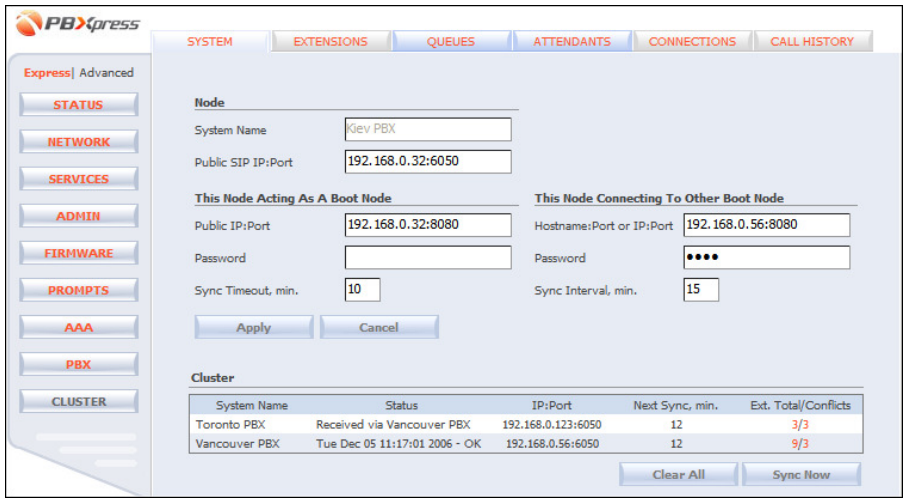
The Node Connecting To Another Boot Node

This allows you to set up this PBXpress as a slave node within the cluster.

Field	Description
Hostname : Port	Connection parameter of the boot node.
Password	Password used to connect to the boot node.
Sync Interval, min.	Re-sync interval (frequency of connecting to the boot node to send/receive updates).

Cluster status

After you submit the information and the nodes exchange the information, you will be able to see the cluster status at the bottom of the screen.



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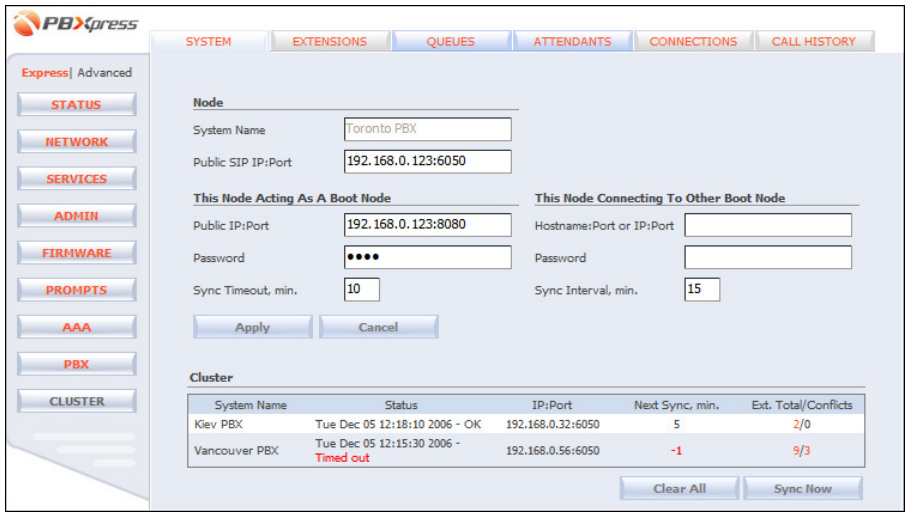
Field	Description
System Name	Unique name of a node
Status	For nodes directly connected to this node, displays the date and time of the last synchronization. For other nodes, displays the name of the node which relayed the information.
IP:Port	Public SIP address and port of this node (incoming calls should be routed to this address).
Next Sync, min.	Time remaining until the next synchronization
Ext. Total /Conflicts	The total number of extensions provisioned on this node and the number of conflicting extensions (the same extension is defined on some other node in the cluster)

You may request the immediate synchronization with the boot node by pressing the **Sync Now** button.

If the synchronization was not performed at the scheduled time, you will see "Timed out" in the node's status and the **Next Sync** column will show the negative number – the number of minutes since the planned synchronization. After the synchronization timeout (defined in the **Sync Timeout** field) is reached – this node is considered to be out of the cluster and information about it is removed from the table.



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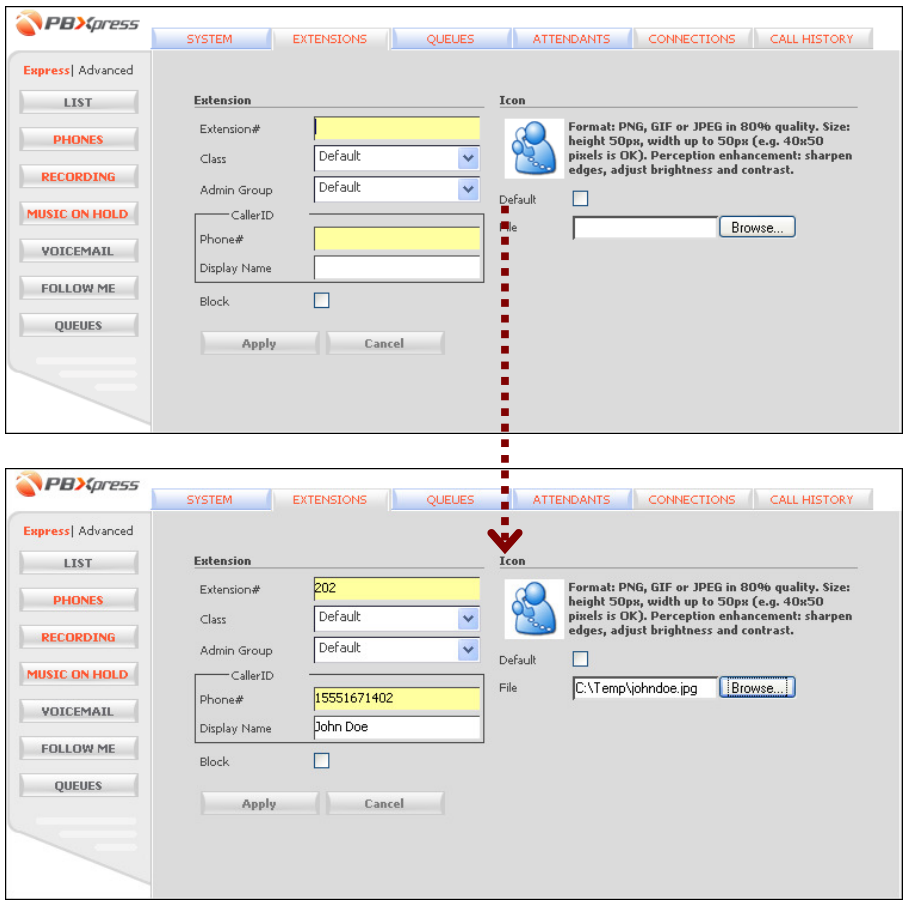
Phone Extension Configuration

Extensions

Extensions are abbreviated numbers for your IP phones. Thus, when someone wants to call John, they can just dial 121 instead of his full number. Here you can manage the list of available extensions – add new entries and edit or delete existing ones.



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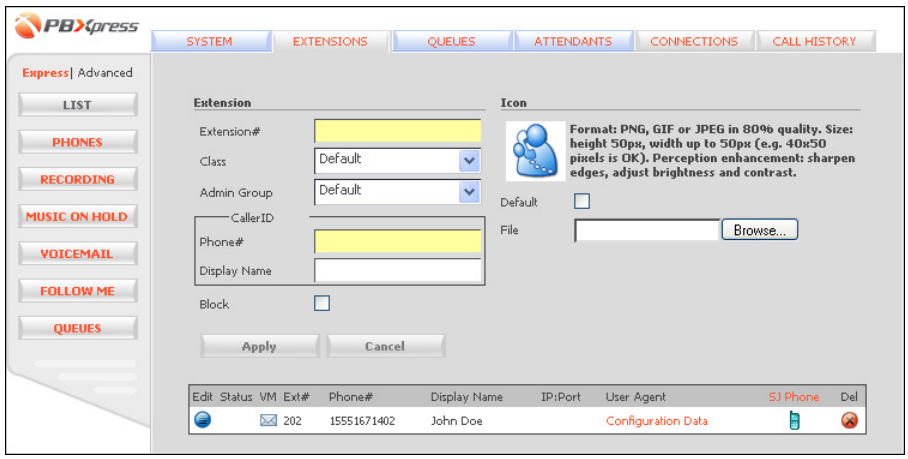
Field	Description
Extension #	Abbreviated number (extension) for this user.
Class	Assigned extension class (defines dialing rules and operations allowed for this user).
Admin Group	Defines a logical "sub-division" within the company PBX environment, for instance a department. The primary role of an admin group is to give an attendant using the operator console access to only a limited set of extensions.
Phone #	The "full" phone number assigned to this user.
Display Name	Identification of the person using the phone. When the user makes a phone call, this string is displayed on the recipient's phone display.
Block	Block a specific extension from making or receiving phone calls.
Icon	Upload a custom picture for easy identification of a user on the interface (especially the operator console).

Phone numbers for all extensions must be unique. So if you do not have "real" phone numbers allocated to all of your users, it is a good practice to assign a phone number which combines your main access number followed by the extension number. For example, if your access number is 18005551234, the extension 201 will be assigned the phone number 18005551234201.

After you have saved the extension data, you will see it in the list.



You need to create an extension for every person in your organization who will use an IP phone.

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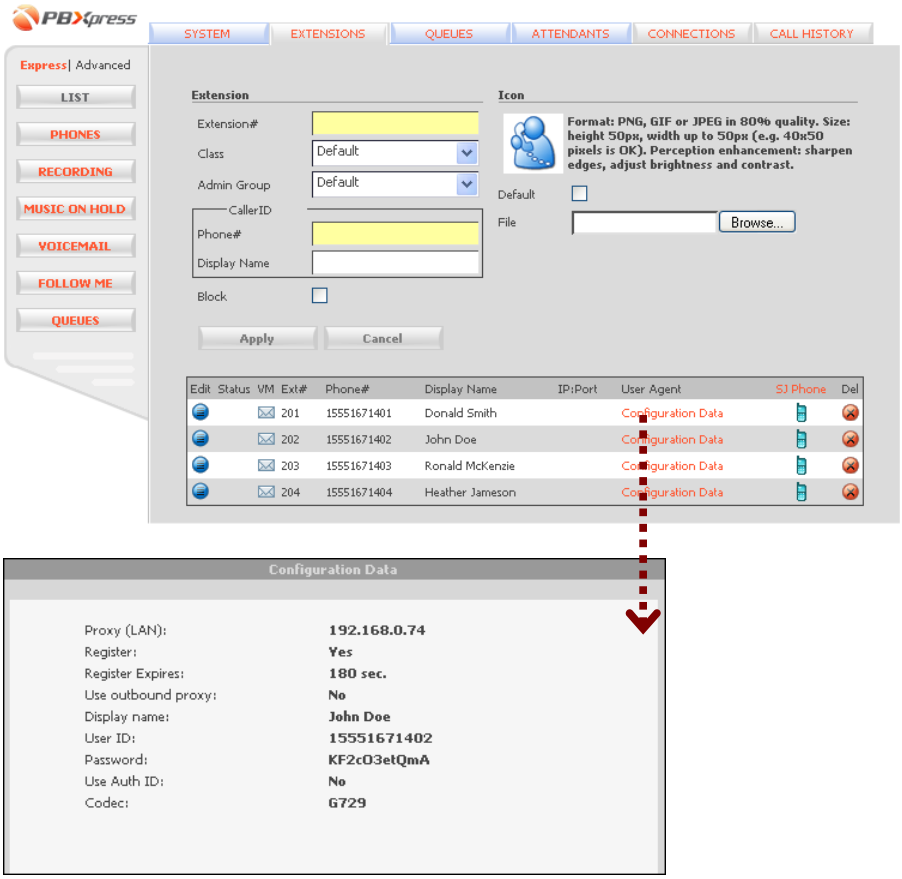
Until you provision extensions to your IP phones, your extensions list will look like the one below. Note the Configuration Data link – this provides you with all the important parameters for provisioning an extension to an IP phone manually, without PBXpress auto-provisioning.



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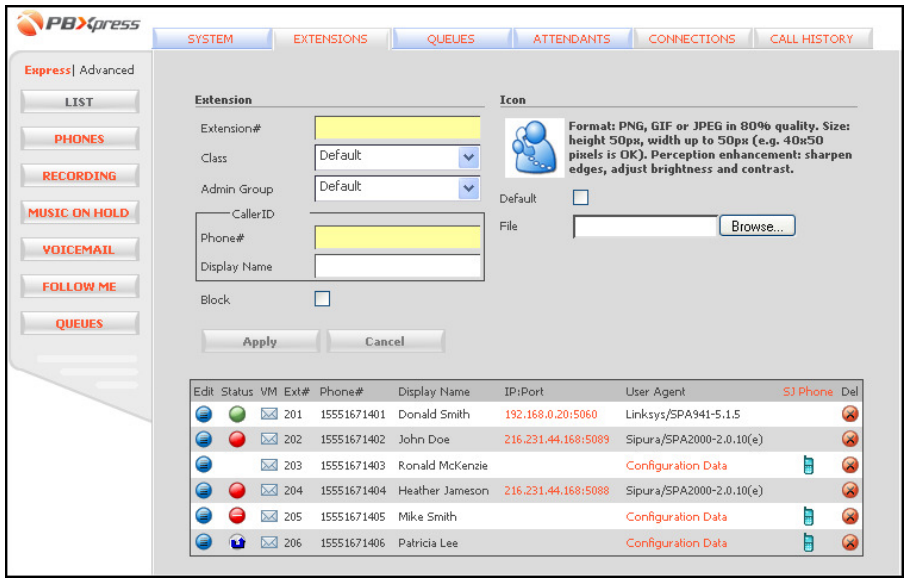
List of extensions



Column	Description
Status	• Empty – this extension is not assigned to any phone, or the phone with this extension is currently offline. • - this extension is blocked. • - this extension is currently busy (a call is in progress) – see below for additional information. • - this extension has been provisioned and the phone is currently online, so this extension can send and receive calls. • - this extension has an unconditional follow-me activated, so all phone calls will be diverted to the external numbers
VM	• Empty – voicemail is disabled for this extension. • – voicemail is activated (by default it is activated for every new extension).
Ext #	Abbreviated number (extension) for this user.
Phone #	Complete phone number for this user.
Display Name	Name of the user.
IP:Port	For a phone which is currently online, the IP address and port of the phone.
User Agent	Type of IP phone currently used.
SJ Phone	SJPhone configuration column. Click on the column header to download and install the SJ Phone on your computer.

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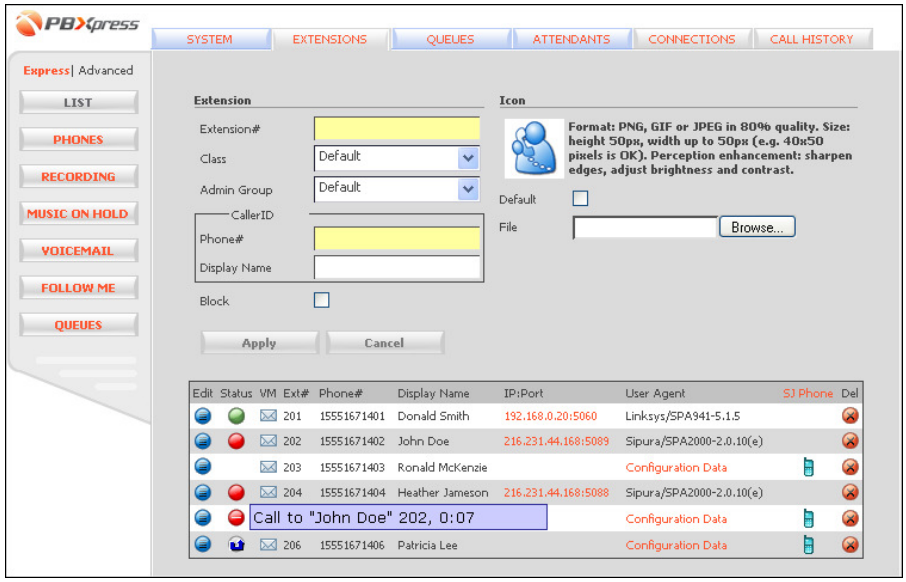
	• - Downloads the SJ Phone configuration file for this extension. After your SJ Phone processes this file, it will automatically be configured for PBXpress. • - Downloads an SJ Phone configuration file which will remove this extension's data from the SJ Phone configuration.

If an extension is reported "busy" you can get more information about the call in progress. Move the cursor over the icon and you will see the destination number and the elapsed call duration, as shown on the picture below.



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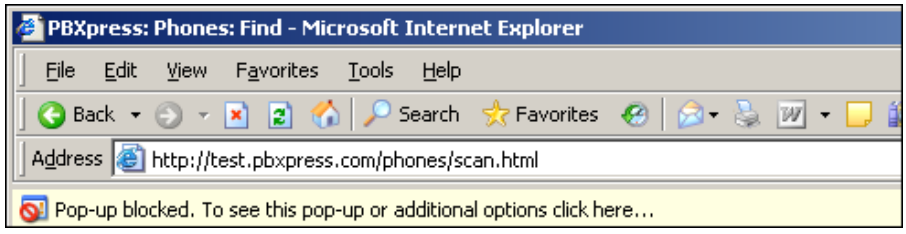
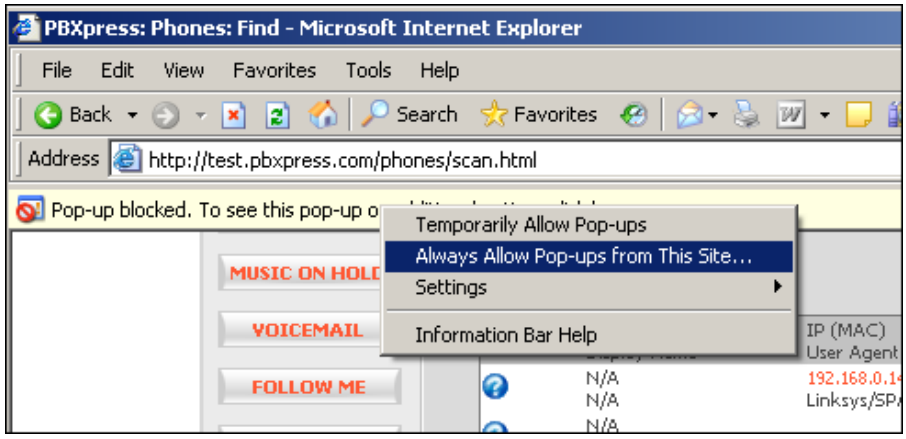
Browser configuration for pop-ups (Internet Explorer)

In order to ensure that auto-configuration works properly, please make sure that pop-ups are allowed for this website in your browser.
You may see a message similar to the one below:

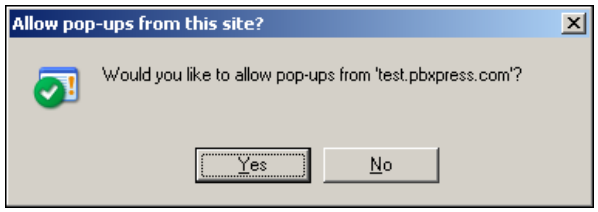


If so, right click on the warning text and choose "Always Allow Pop-ups from this Site":



Browser configuration for pop-ups (Mozilla)

Mozilla/Netscape users may see a message similar to the one below. Click **No**, so that pop-ups will be allowed.



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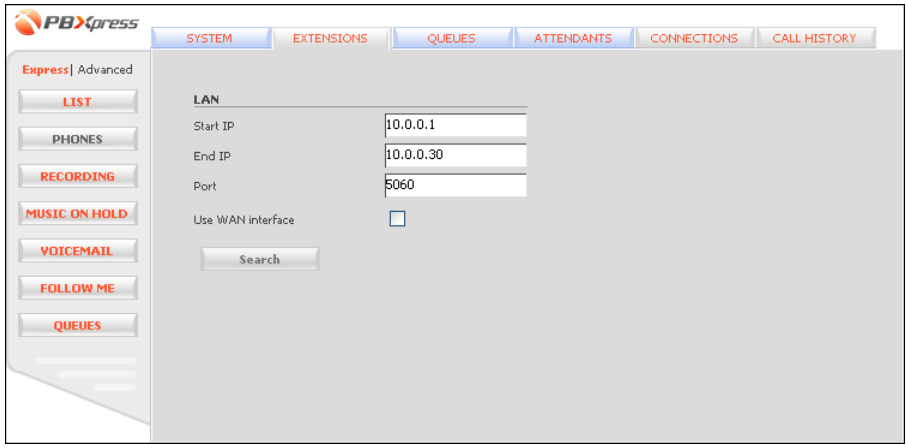
Phones

Usually, when putting IP phones on your network, you need to configure each one of them individually before they can be used. PBXpress allows you to substantially simplify this process. You can now simply connect IP phones to your LAN in the default factory configuration. PBXpress will detect and configure all of them automatically.



Specify the starting and ending IP (in most cases, the default values provided are OK) and press **Search**.

Normally, IP phones listen to incoming connections on port 5060 (the standard SIP port). However, if you have specially pre-configured phones you may specify a different port value.

By default, the search is performed on the LAN, but you may also search for phones on the WAN. When using this mode, be careful to specify as narrow an IP range as possible. Also note that in order to be detected on the WAN, an IP phone must be currently connected directly to the Internet (i.e. on a public IP and not behind a firewall or NAT).

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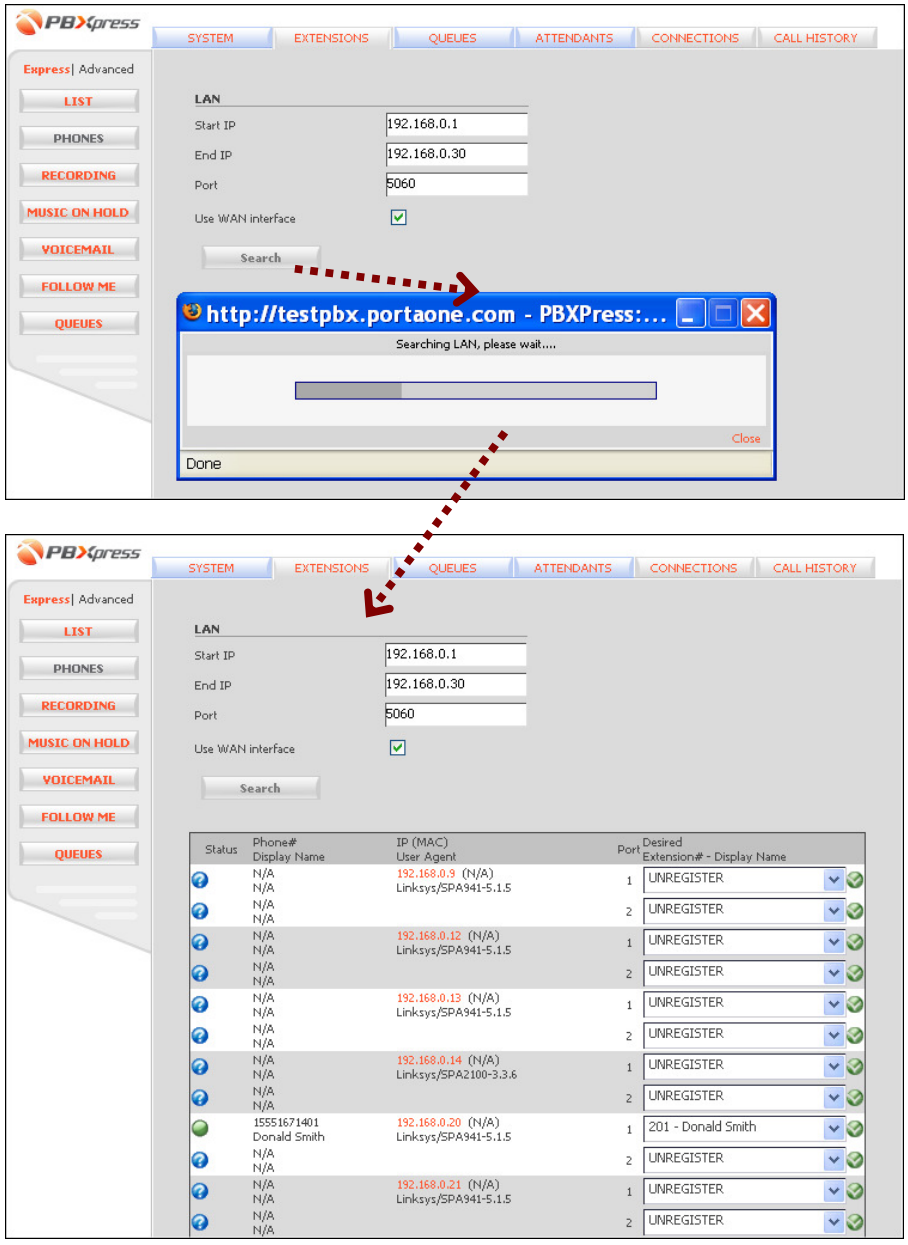

You will see a list of the IP phones which were detected on the network. Note that since these phones are not yet configured, they are marked as unregistered. Now you can assign one of the phone extensions to the IP phone (or to a specific port on it). PBXpress will update the phone configuration so that it is registered to your server.

Select the extension that you would like to provision on a specific phone, and click the icon.

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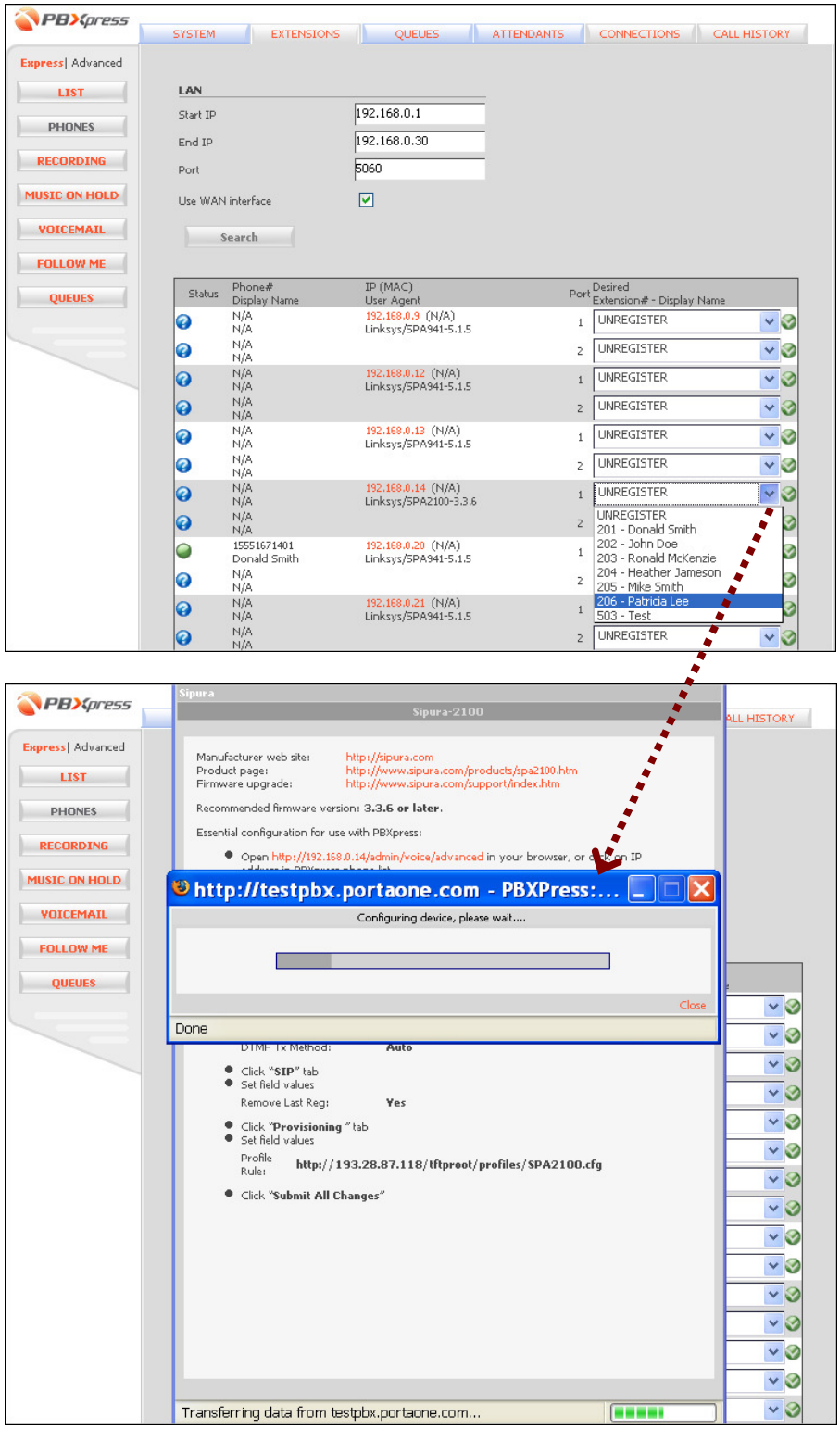

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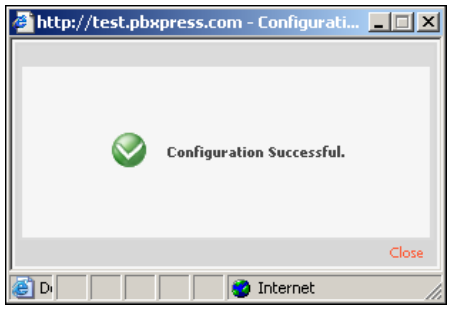
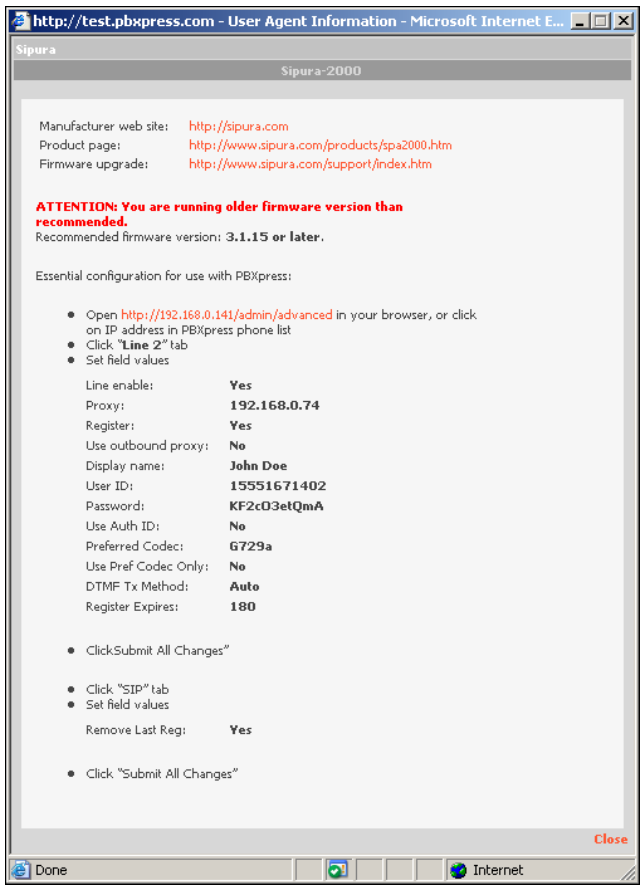
When phone configuration has been completed, you will see a confirmation message similar to this one:



Press the **Close** button to close the pop-up window and return to the phone configuration screen. PBXpress will re-scan the network and display the current status of the phones.

Note that another pop-up window will open during IP phone configuration. This window will contain additional information about the IP phone model. You will see an information screen similar to the one below.



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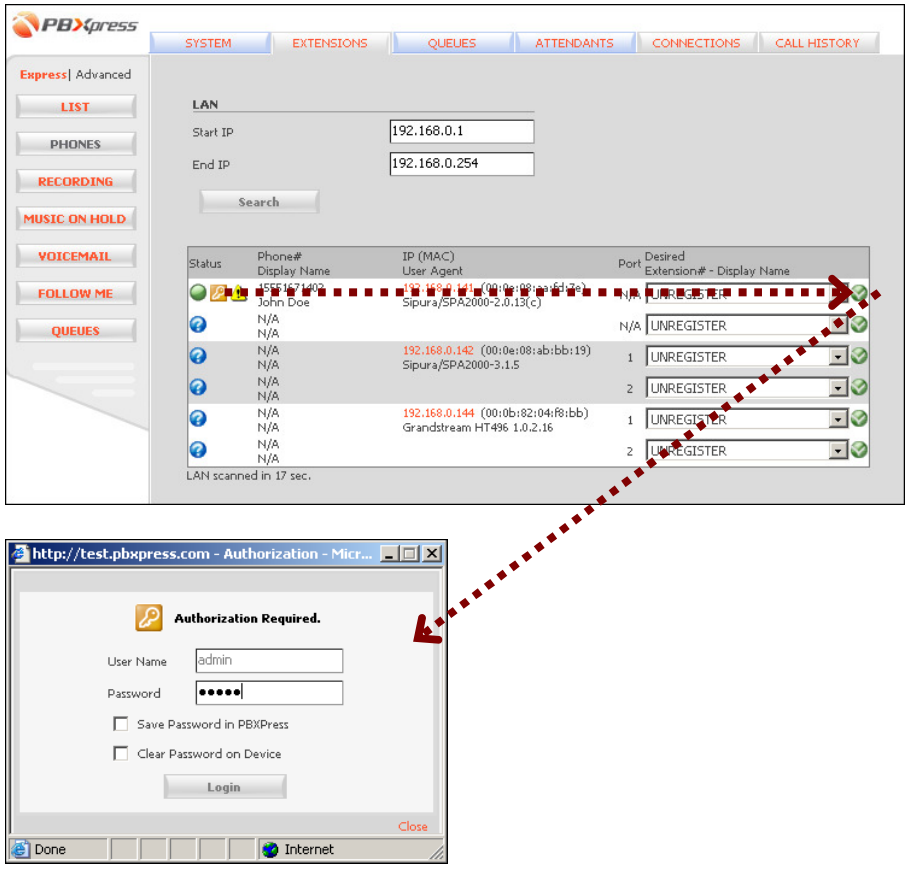
Should configuration of the IP phone be password-protected, you will be prompted to enter the password (you can also store it in PBXpress for future use).



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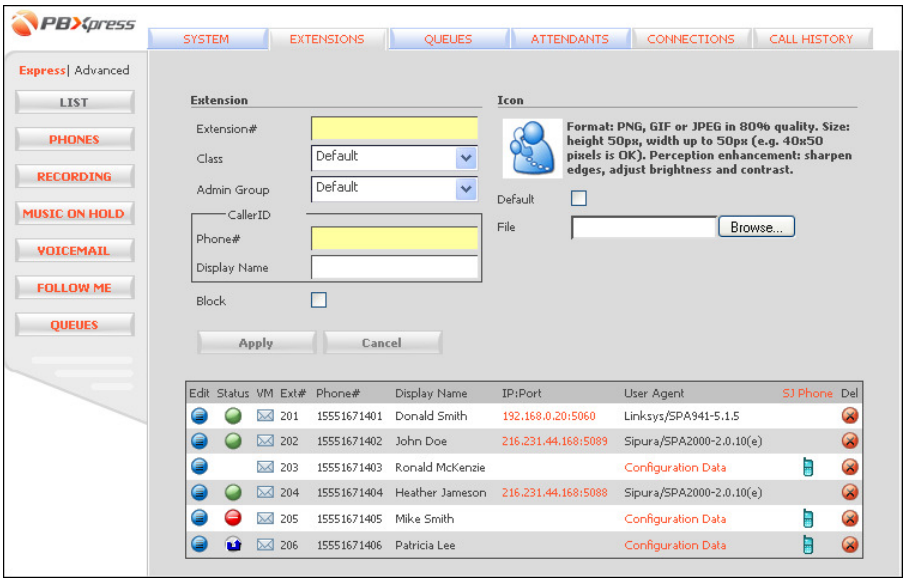
After you have configured several IP phones, you can go back to the list of extensions (click the **List** button on the left). Extensions which are currently registered to the server will have a green status light :



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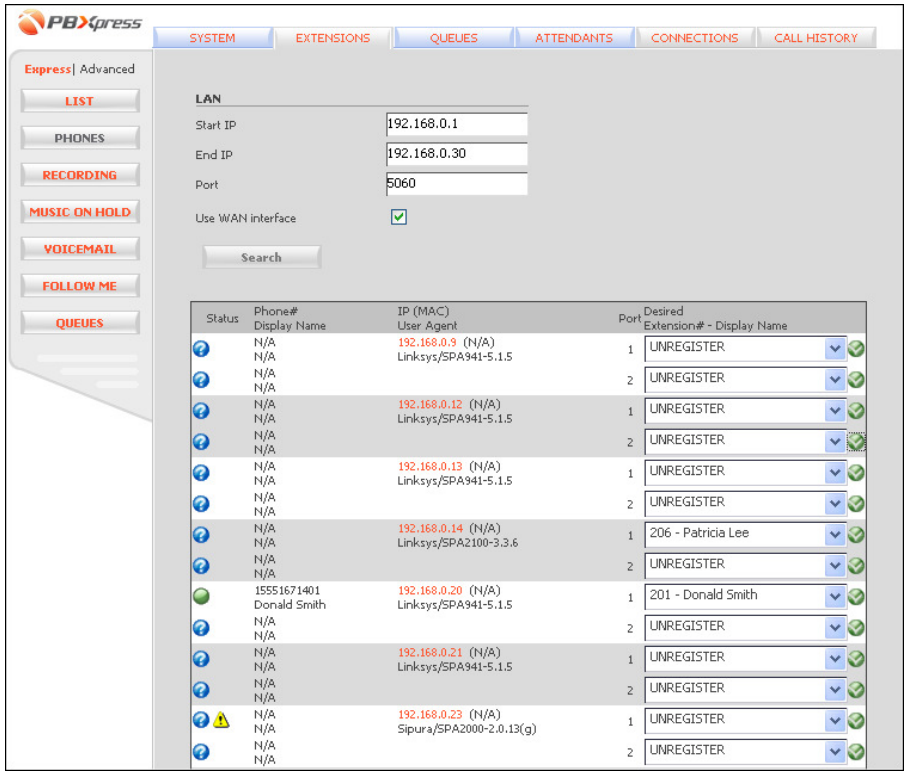
IP phone firmware version

If, during IP phone detection, PBXpress detects that one of the phones is currently using outdated firmware (which may have known bugs, or may not have all the required features) you will see a warning icon, as shown below.



Please consult the IP phone information page (the pop-up window which appears during phone configuration) to find out which firmware version is recommended.

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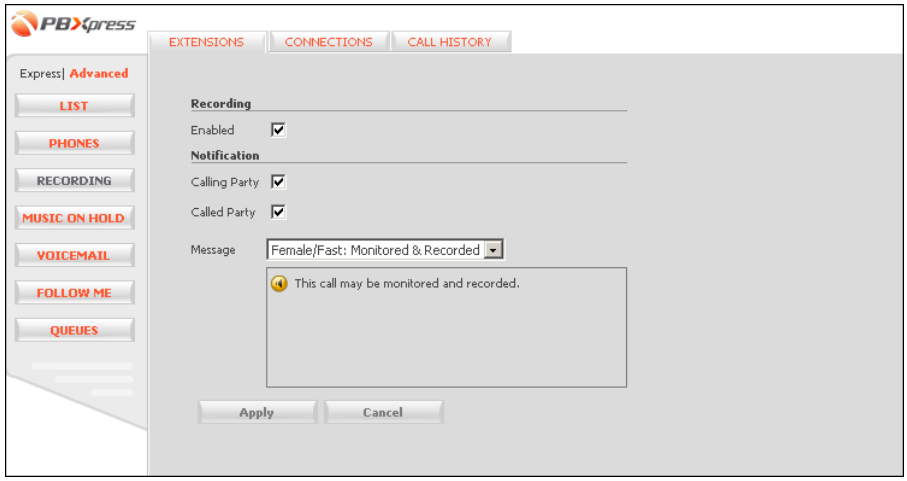
Recording

You can turn call recording on or off for calls between different IP phones on your network (the default is on), also choosing who will be notified about the fact that a call is being recorded and which announcement (of those defined in the **System/Prompts** section) will be used.



Field	Description
Enabled	Turn call recording on or off (for calls between extensions).
Calling Party	Notify the caller before the conversation starts that the call will be recorded.
Called Party	Notify the recipient of the call.
Message	Which voice prompt should be used for the call recording announcement.

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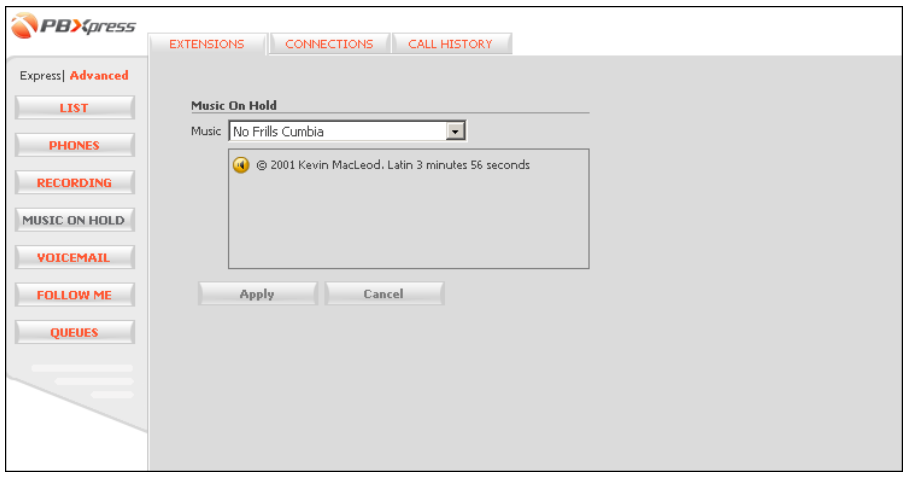
Music on Hold

This screen defines what music should be played if one party is put on hold during a call between two extensions. Choose one of the pre-defined voice prompts (**NO MUSIC** means that nothing will be played) and press **Apply**.



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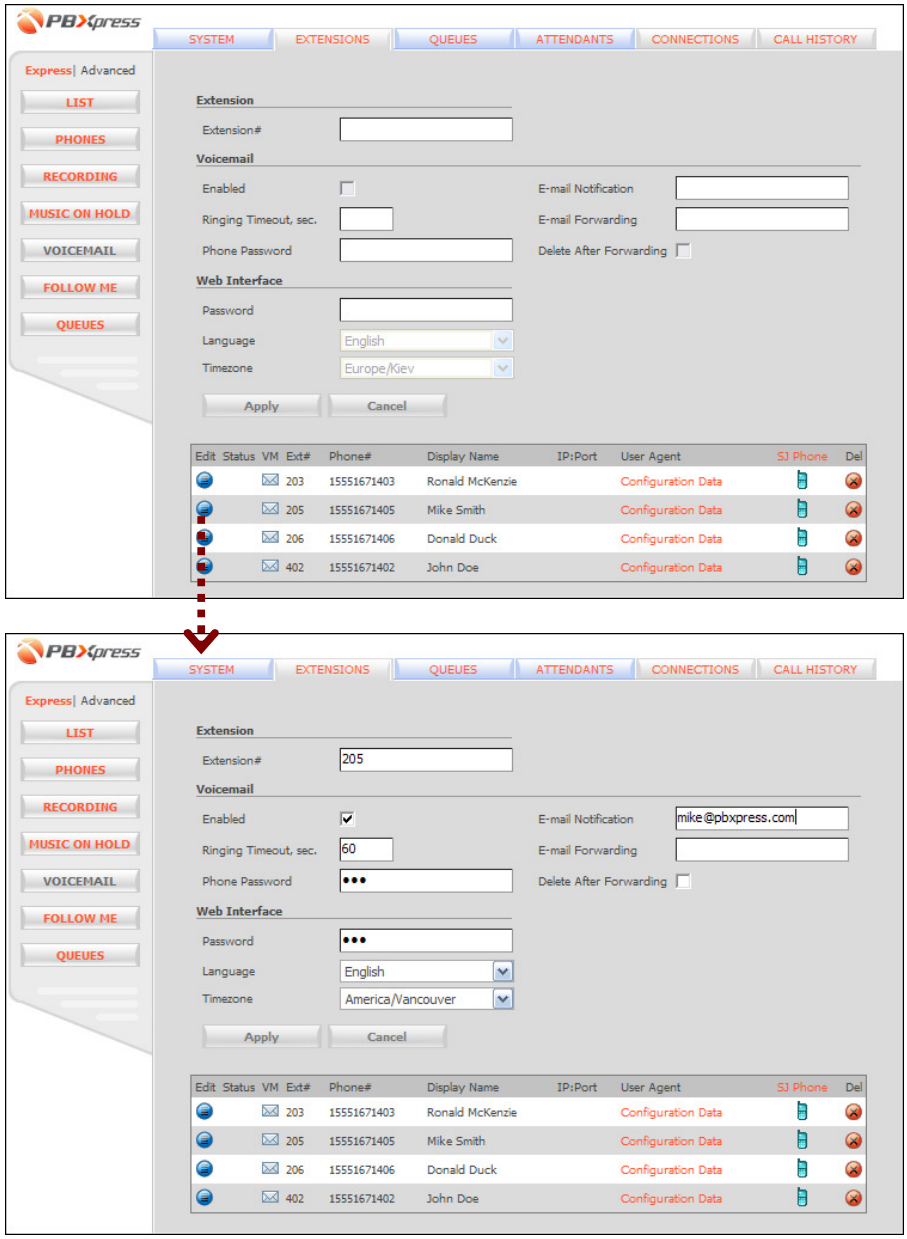
Voicemail

When an incoming call to an extension is not answered, it will be forwarded to voicemail so that the caller may leave a message.



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Field	Description
Extension #	The extension number.
Enabled	Activates voicemail for this extension.
Ringing Timeout, sec.	How long the phone will ring before the call is transferred to voicemail. (By default the value defined in the **Ringing timeout** in the **System/PBX** properties is used).
Phone Password	Password for accessing voice messages from an IP phone (by dialing the voicemail access number). By default, the password is identical to the extension number. If you enter an empty password, the system will not ask for the password when the voicemail access number is called from an IP phone.
E-mail Notification	When a new voice message arrives, a short notification is sent to the specified email address.
E-mail Forwarding	Sends a complete copy of the every new voice message to the specified email address.
Delete After Forwarding	Do not leave a copy of the message on PBXpress when forwarding a message to an external email address. In this case, you will not be able to listen to this message from your IP phone or via the PBXpress web interface.
Password	Password for accessing voice messages via web interface. By default, the password is identical to the extension number.
Language	Specifies which language is used for this extension's self-care web pages.
Timezone	To display call times in the specified time zone on the extension's self-care pages.

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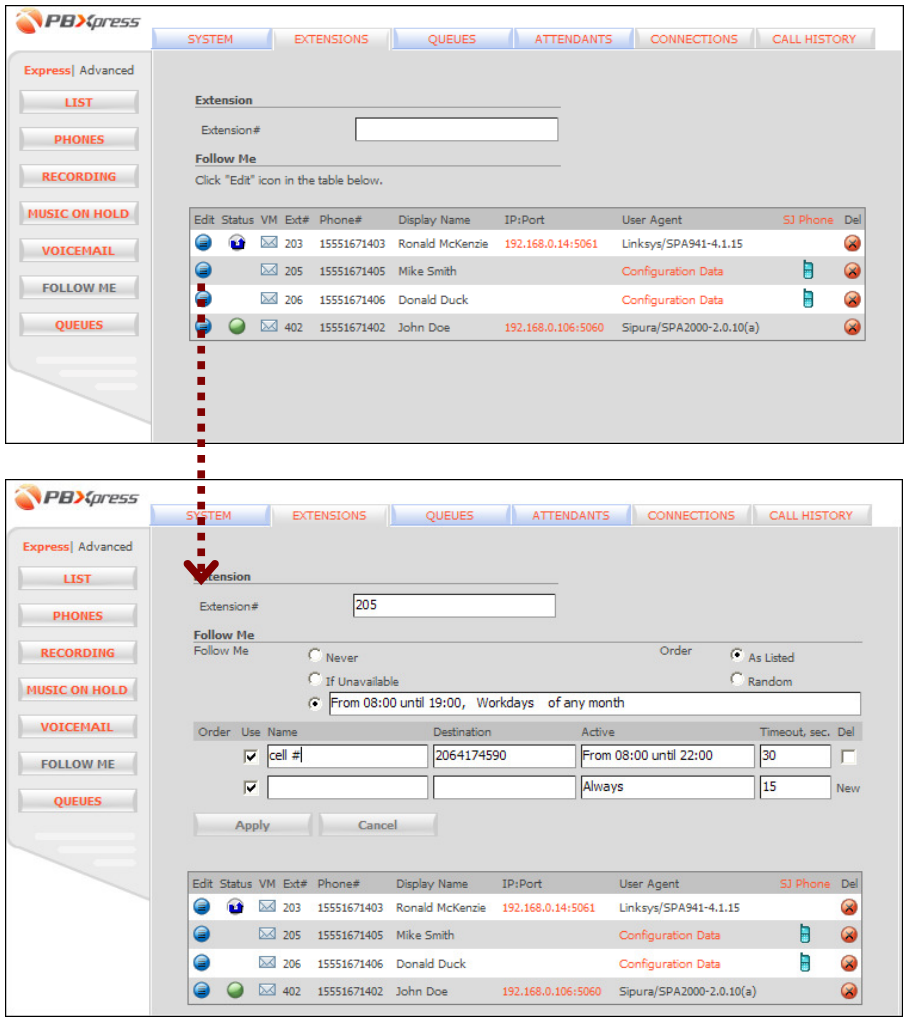
Follow-me

When there is an incoming call to an extension, one of the following may happen:

- The extension is offline (not registered).
- No one answers within a certain time interval.
- The extension is currently engaged in a phone conversation and has "Call forward on busy" activated.

In these cases, it is impossible to deliver the call directly to the IP phone. The call may then be forwarded to another extension, or to a PSTN phone (mobile, office, home, etc.).



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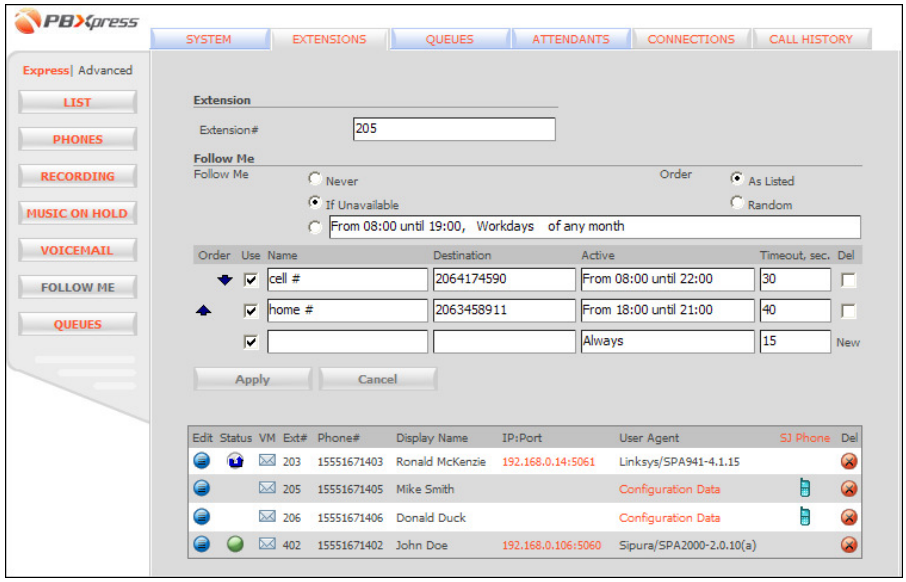
follow-me parameters:

Field	Description
Follow Me	• **Never** – Turn off the follow-me functionality. Useful if you want to temporarily disable follow-me for this particular account but keep all the other settings, such as the list of forward phone numbers. • **If Unavailable** – First try to ring on the IP phone. If not answered within a certain number of seconds, go to the follow-me numbers. • The last option allows you to specify a time period. Within the specified interval, PBXpress will not even try to ring on the SIP phone, but will immediately forward the call to one of the follow-me numbers. Useful when you know that you will not be able to answer your IP phone at a certain time (e.g. at night, or when leaving on vacation).
Order	• **As Listed** – Ring on the follow-me numbers sequentially, in the order they are listed on the web interface. • **Random** – Shuffle the list of numbers each time to produce a new sequence.

You can enter multiple follow-me numbers; see a description of number parameters in the table on the next page.



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follow-me numbers configuration:

Field	Description
Order	Use the arrow buttons to change the order of the follow-me numbers in the list.
Use	Enables quick disabling of forwarding to this particular destination, but keeps all other information (phone number, period) intact. Useful when you do not want to use a phone (e.g. your overseas cell phone) right now, but may need it again in the future (during your next business trip).
Name	Easy-to-understand identification of the destination (e.g. My Mobile).
Destination	Phone number the call should be forwarded to. The number must be entered in exactly the same format as it would be dialed from an IP phone. Thus, if you want to forward the call to extension 121, enter "121"; if you want to forward it to your overseas mobile, which you would dial as 011420732813200, then that is the number you should enter.
Active	By default, forwarding will work 24 hours a day (**Always**). If you wish to change this, click on the value in this column. The period wizard will appear, allowing you to precisely define when calls should go to this number.
Timeout, sec.	How long PBXpress should wait for someone to answer this phone number before moving on to the next number in the follow-me list.

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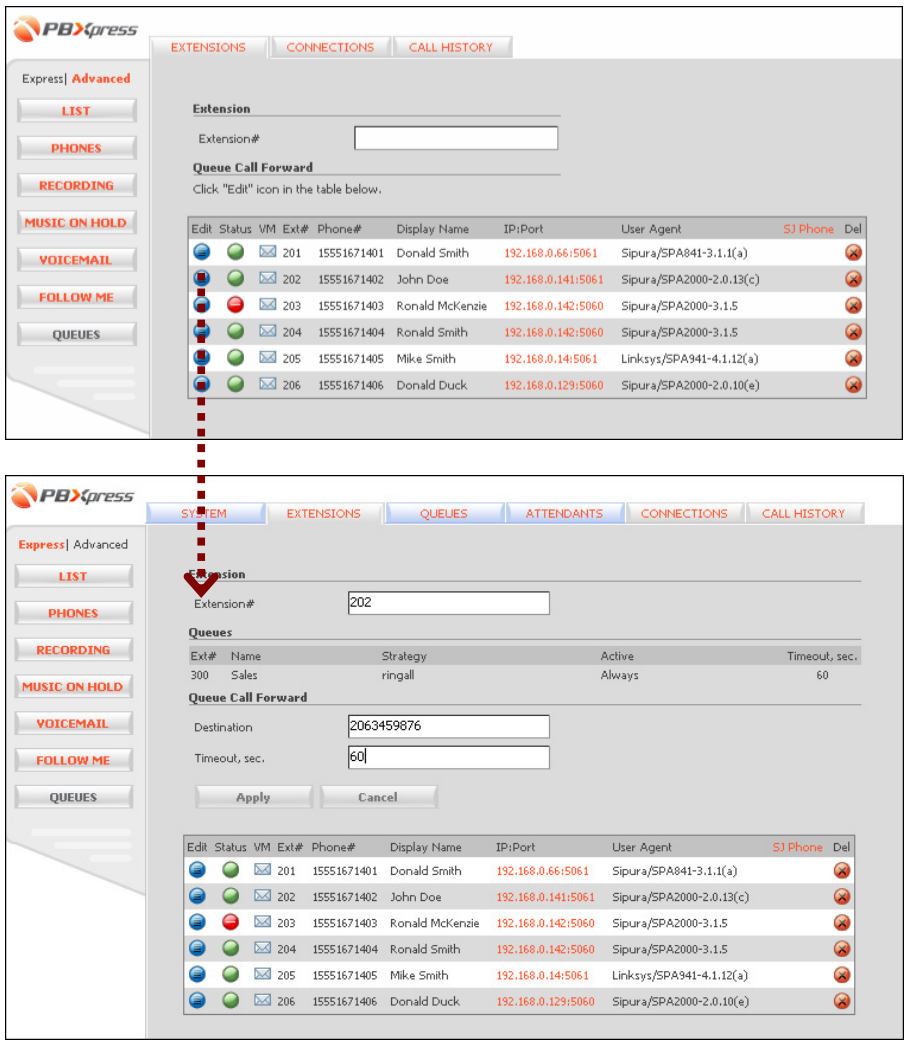
Queues

This section specifically controls calls which arrive to an extension from a queue (select an extension in the **Queues** table to see what queues it is part of).



NOTE: When this parameter has been set up, the extension will not receive any incoming calls from queues; rather, all calls will be automatically diverted to the specified number.

Field	Description
Destination	The phone number a call should be forwarded to. The number must be entered in exactly the same format as it would be when dialed from an IP phone.
Timeout, sec.	How long PBXpress should wait for someone to

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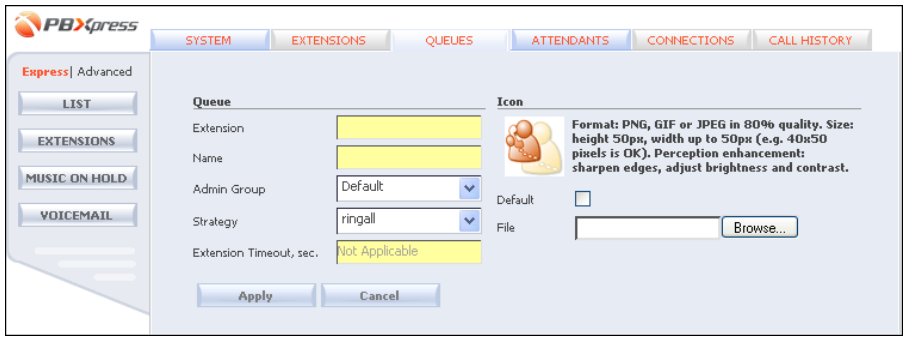
	answer this phone number.

Queue Configuration

Create a queue

Click on the **Queues** tab. You must create at least one queue, since initially there are no queues available.



Fill in the queue parameters, then press the **Apply** button.

Field	Description
Extension	Extension number to reach this queue from an IP phone. The number will be reserved for this queue, and cannot be used for any actual IP phone.
Name	Name of the queue.
Admin Group	Defines a logical "sub-division" within the company PBX environment, for instance a department.
Strategy	How the incoming call is distributed between the available extensions. See the *Queues* section for detailed information about the different strategies.
Extension Timeout, sec.	How long an incoming call will wait to be connected (not applicable to the **ringall** strategy).
Icon	Upload a custom picture for easy identification of this queue on the interface (especially the operator console).

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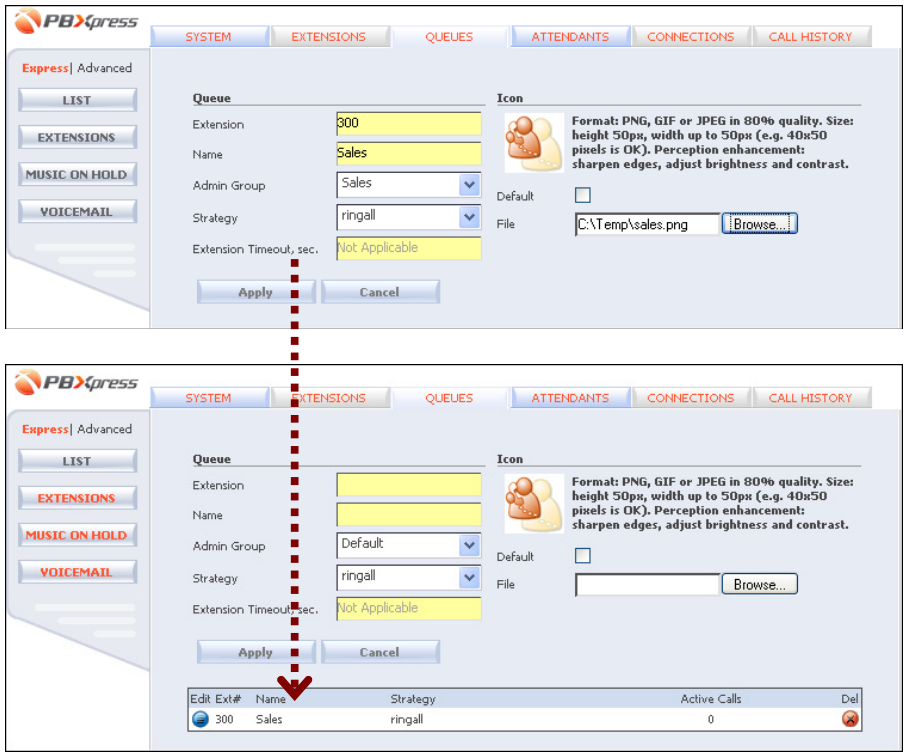

You will see a list of the current queues. Note that the queues list can also be used to monitor current calls to a particular queue.

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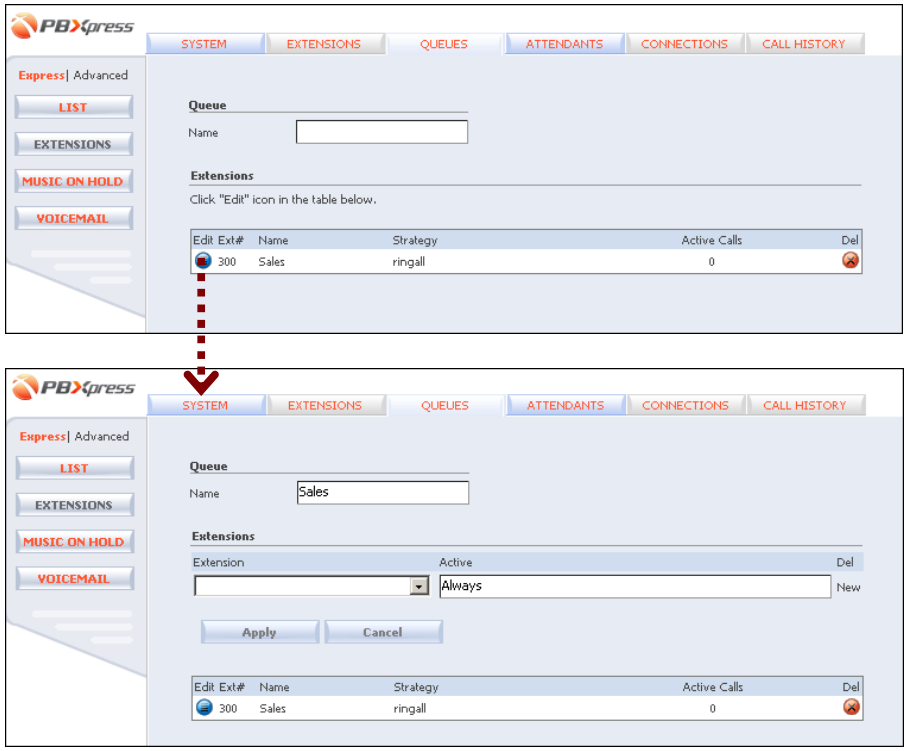
Extensions

After a queue is created, you may edit the list of extensions for it. Press the **Extensions** button on the left and then click on the Edit icon for the specific connection.



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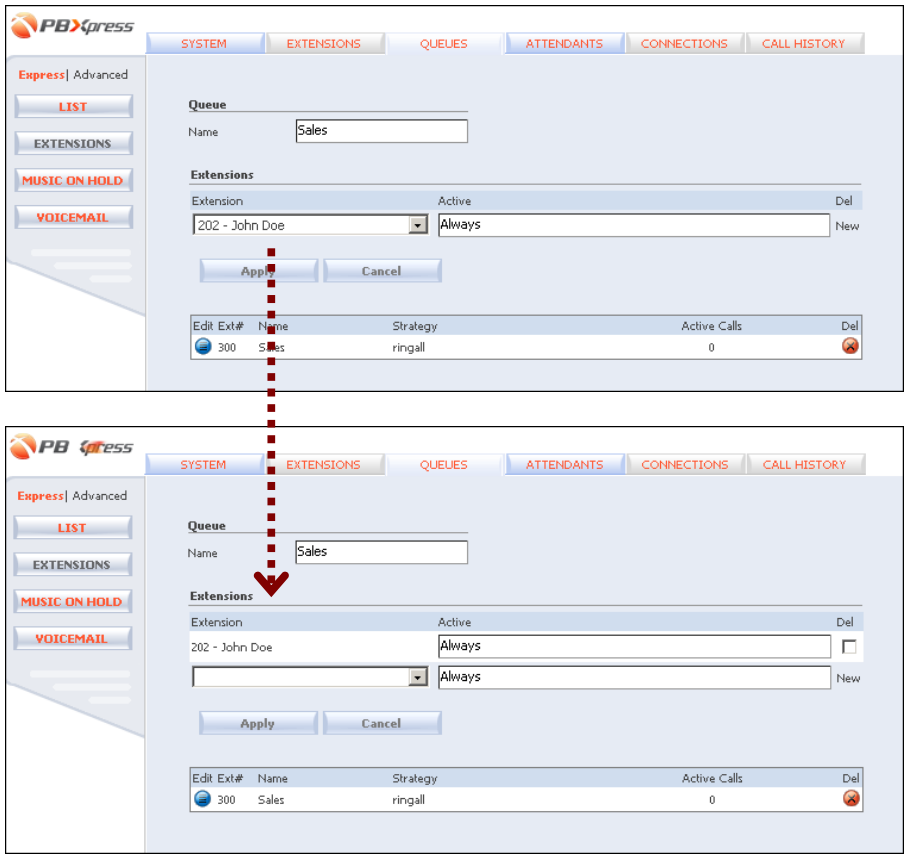
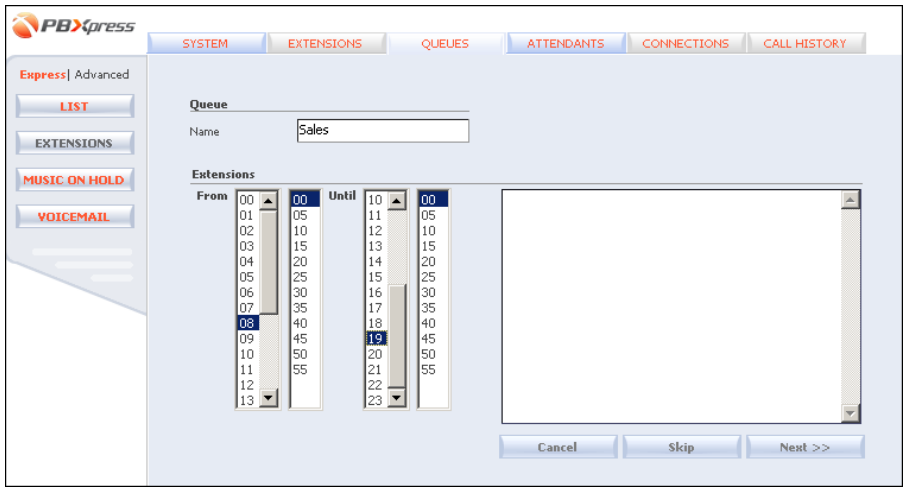
In order to add an extension to the queue, choose it from the Extensions menu and press **Apply**.



You may include a certain extension in the queue only for a certain time interval (e.g. only during office hours). To define the activity period, click on the value in the **Active** column, and the period wizard will appear.



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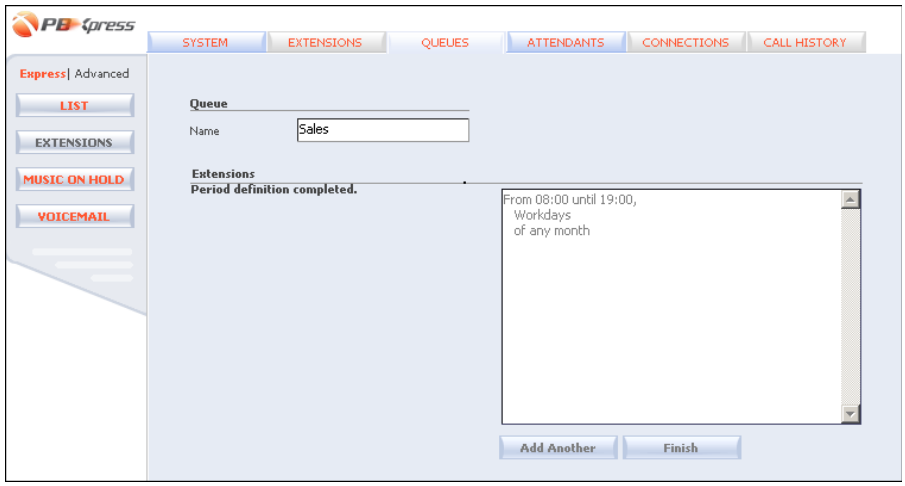
Choose the applicable time constraints on the left and press **Next** in order to proceed.



After the period has been defined, click **Finish** to return to the main extension configuration screen.

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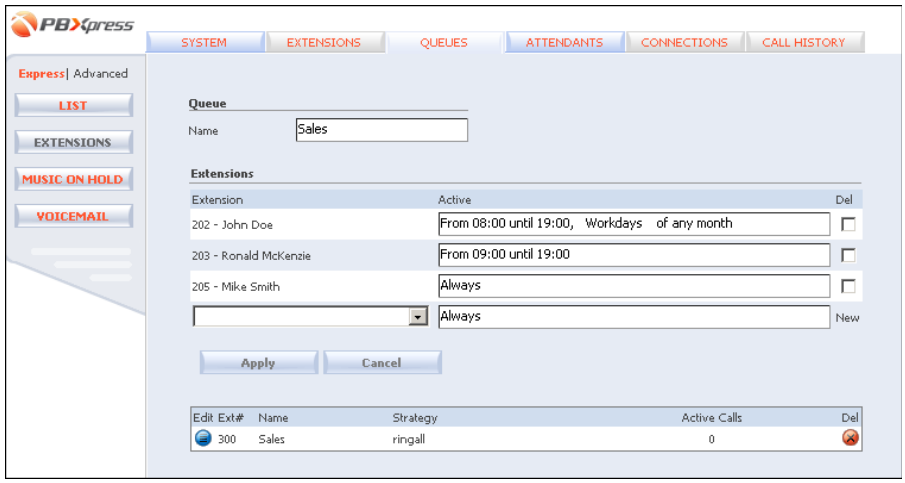
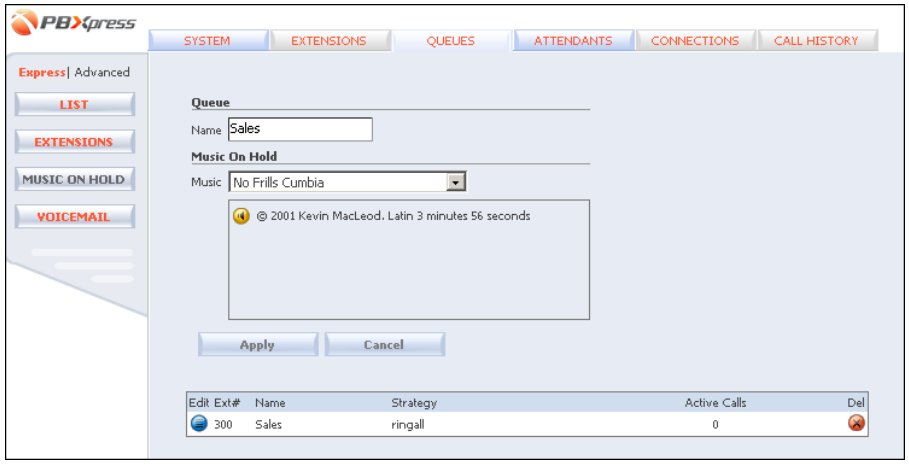
Repeat the above steps to include all the required extensions in the queue, as shown below.



Music On Hold

Here you may select which music a caller will hear while he/she is on hold. You may assign different sound files to every queue.



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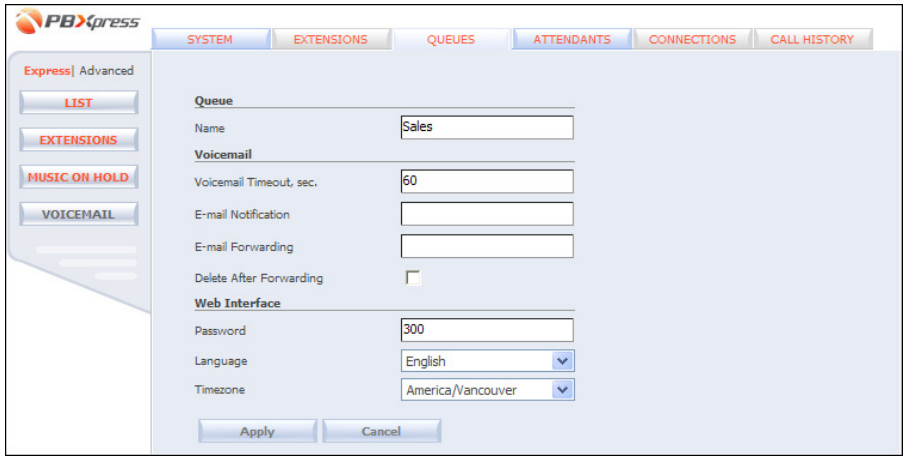
Voicemail

If no one in the queue answers the call, voicemail will be engaged – very similar to what happens with ordinary extensions.



Field	Description
Name	Name of the queue.

Voicemail section.

Field	Description
Voicemail Timeout, sec.	Maximum time to wait for someone in this queue to answer the call.
E-mail Notification	When a new voice message arrives, a short notification is sent to the specified email address.
E-mail Forwarding	Sends a complete copy of the every new voice message to the specified email address.
Delete After Forwarding	Do not leave a copy of the message on PBXpress when forwarding a message to an external email address. In this case, you will not be able to listen to this message from your IP phone or via the PBXpress web interface.

Web Interface section.

Field	Description
Password	Login to self-care interface with the queue extension number and this password
Language	Specifies which language is used for this queue's self-care web pages.
Timezone	To display call times in the specified time zone on the queue's self-care pages.

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Attendant Configuration

In this section you will learn how to program the IVR menus to be used in your IP PBX system. The most typical scenario is when you have one "central" contact number for your organization. When a customer calls it, he should be presented with different options (sales, customer support, etc.) and then forwarded to the correct recipient.
When a call arrives to the attendant, the individual attendant IVR is launched. Then, depending on the user's actions, he may be transferred to another menu (e.g. from the main company directory to the sales department menu – this allows you to create hierarchical IVR trees), or to a specific extension.
Each attendant IVR menu has a set of parameters, which can be divided into three categories:

1. Menu voice prompts – Voice prompts which are played to the user to describe available choices.
2. Schedule – Defines the period when this menu is active. For instance, you can have an extended IVR during office hours, and a very simple voice mailbox during the night.
3. List of actions – What happens when a user presses a specific key.

See the flow chart on the next page for a better understanding of the attendant logic.

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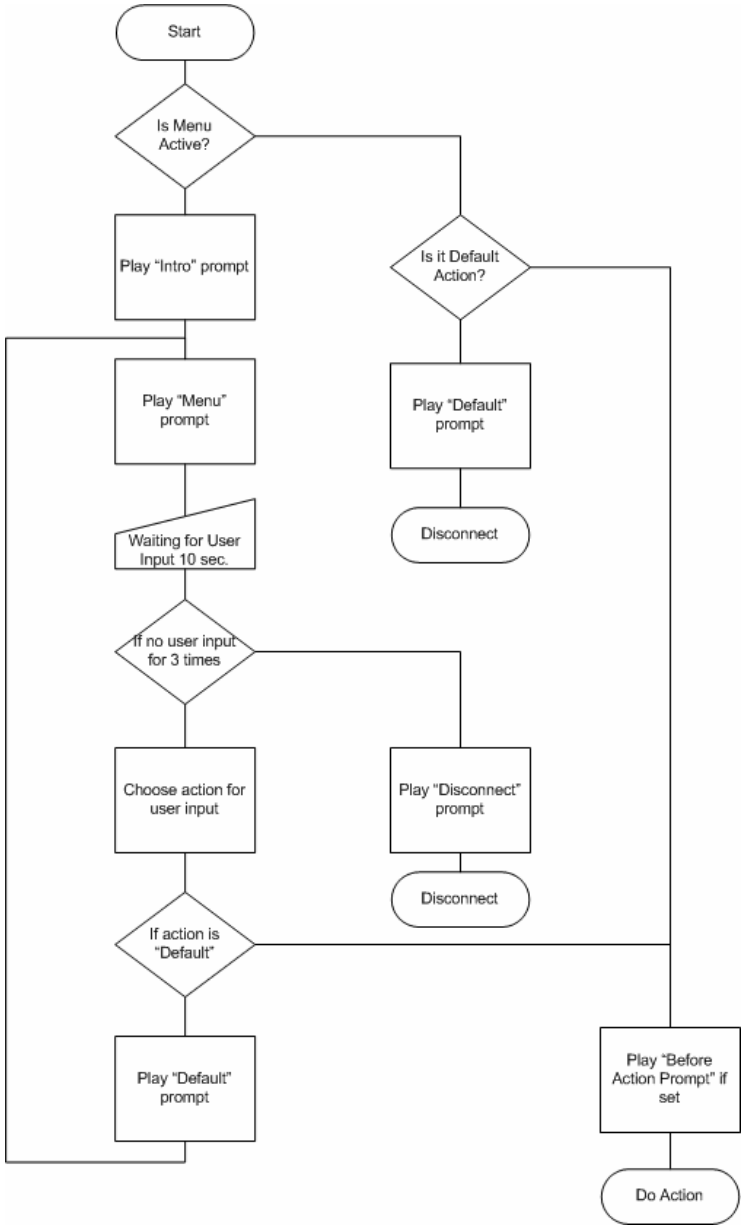

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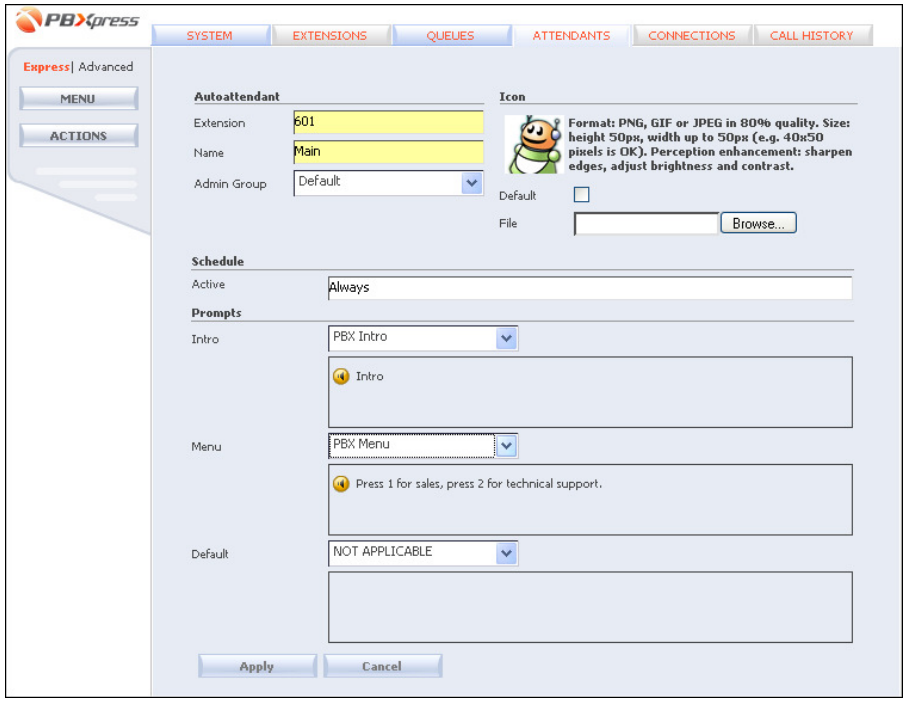
Create a new attendant menu

Hint: Before starting to design your attendant IVR menus, make sure you have recorded all the required voice prompts and uploaded them to PBXpress via the System/Prompts menu.



Available fields:

Field	Description
Extension	The extension number to reach this attendant from an IP phone. The number will be reserved for this attendant, and cannot be used for any actual IP phone. It is a good idea to group all the attendant menus within the same extension group, e.g. 601, 602, etc.
Name	The name to be assigned to this menu. This name will be used to identify the IVR attendant menu when assigning it in the **In routing** connection parameters.
Admin Group	Defines a logical "sub-division" within the company PBX environment, for instance a department.
Icon	Upload a custom picture for easy identification of this attendant on the interface (especially the operator console).
Schedule	The time period when this attendant is active. Click on the field to invoke a period definition wizard; see below for more details.

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Prompts parameters:

Field	Description
Intro	Voice prompt to be played when a user first reaches the voice menu within the current IVR session (e.g. "Welcome to PBXpress Communications Ltd").
Menu	Voice prompt explaining the different available options (e.g. "Press 1 for sales, press 2 for technical support, etc").
Default	If no input was received from the user within the timeout interval, or the user pressed a key that does not have an associated action, this prompt will be played and the menu will start again.

You can choose **NOT APPLICABLE** for any of the prompt parameters, in which case that type of prompt will not be played at all.

Create all the other required menus or sub-menus in a similar fashion. For instance, if the main menu contains a "Press 1 for sales" item, a "Sales" attendant menu should be created to host all of the available options (e.g. "Press 1 for computer sales, press 2 for appliances").

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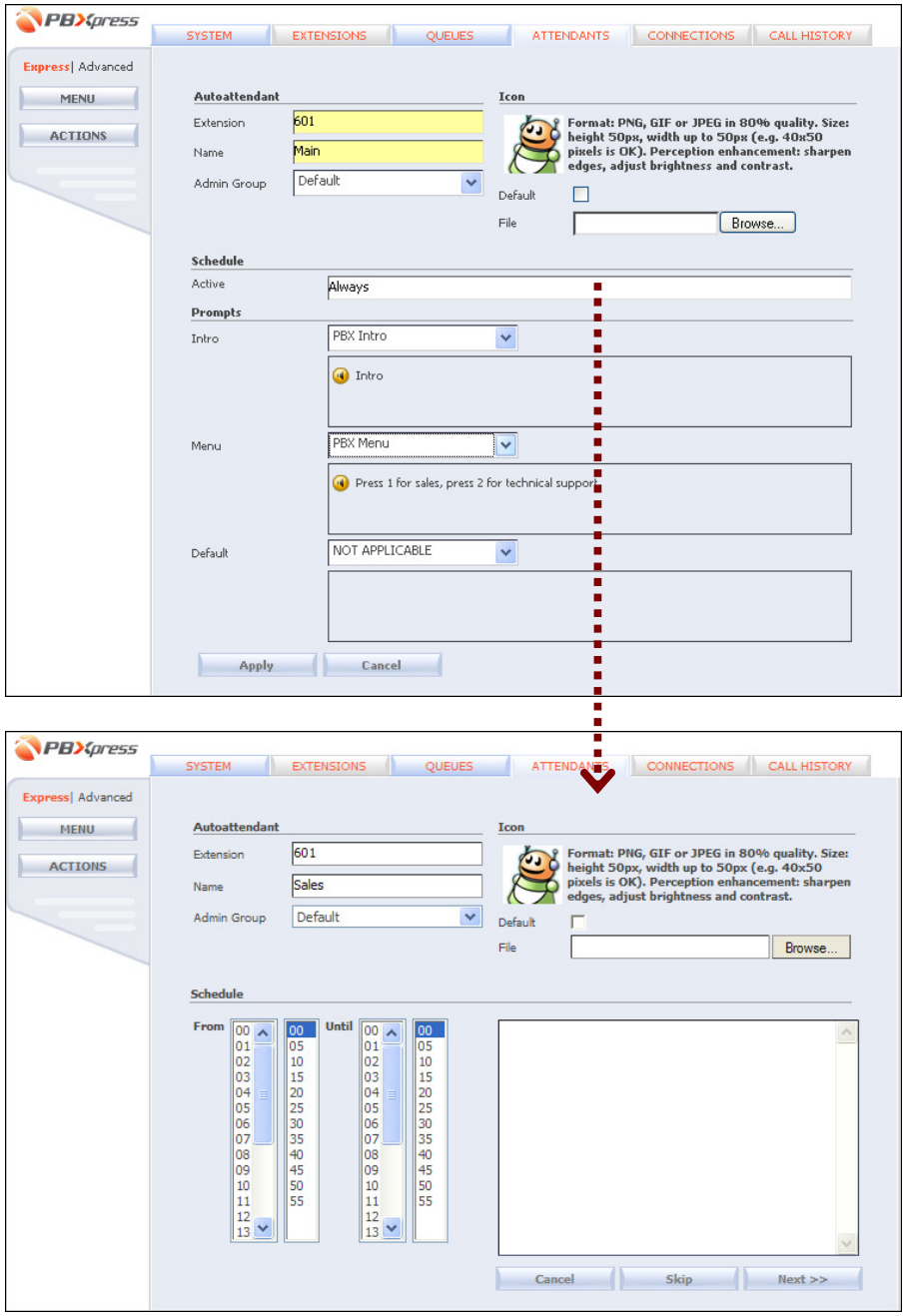
Schedule

Here you can specify when a specific attendant menu will be active. Either choose **Always** (the default option) to activate the menu regardless of the current time, or use a wizard to create a flexible period definition.



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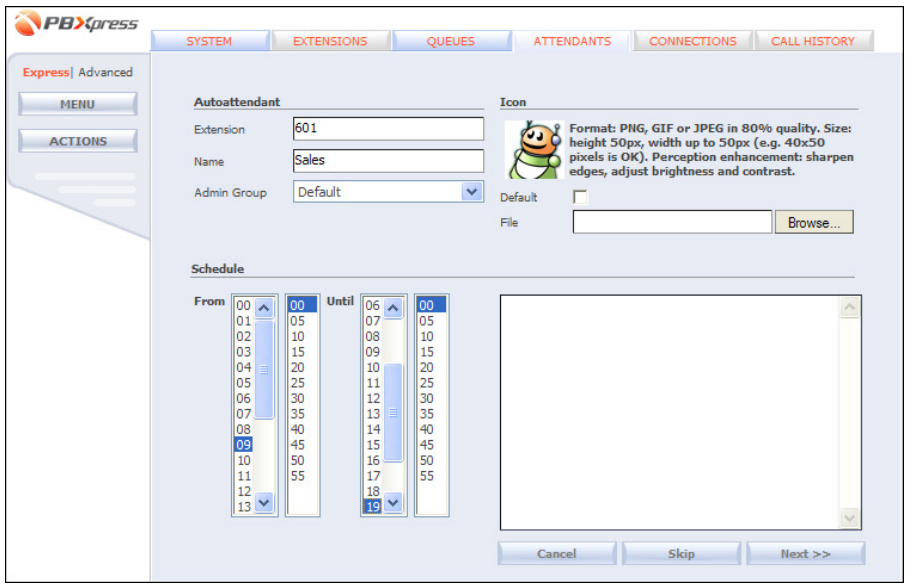
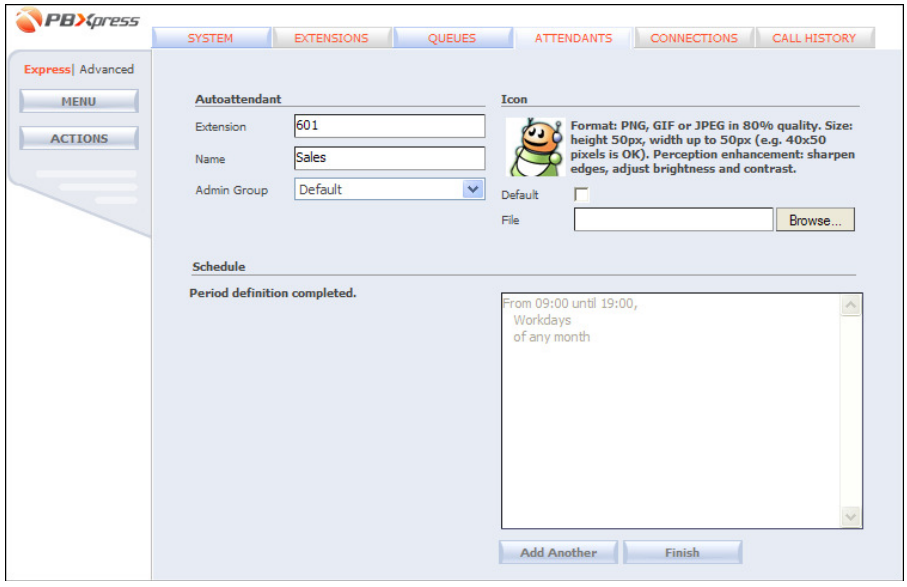
You can specify a period based on a combination of any of the following:

- Time of day
- Day of the week
- Day of the month
- Month

You can also have multiple period definitions, e.g. "9-19 every working day OR 10-17 every Saturday".



If you do not need a limitation by the current parameter (e.g. by the day of the week), press **Skip** to move to the next section.



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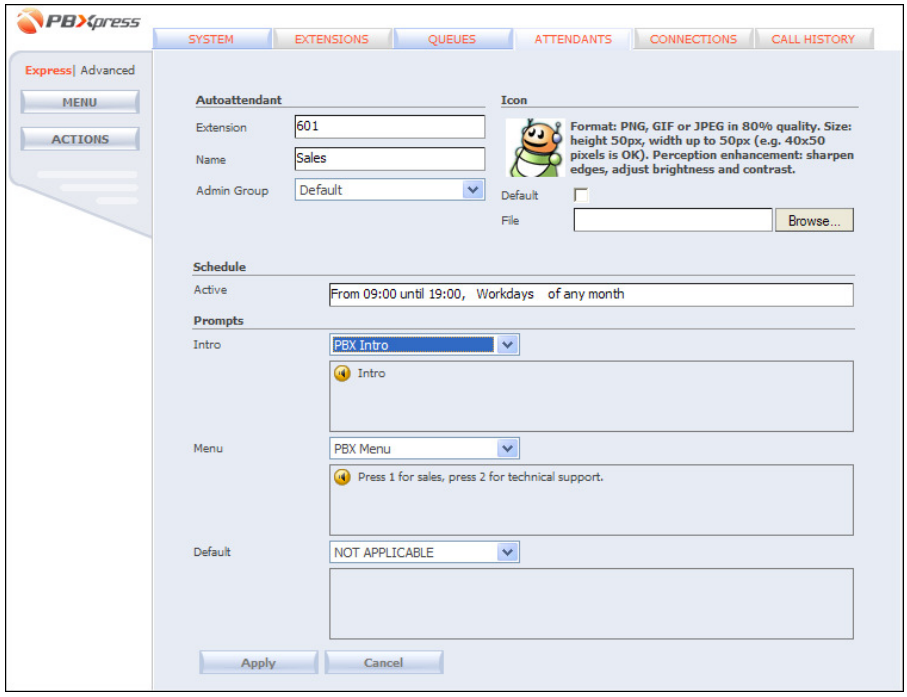
After the definition is complete, press **Finish** to exit the wizard and then **Apply** to save your changes.



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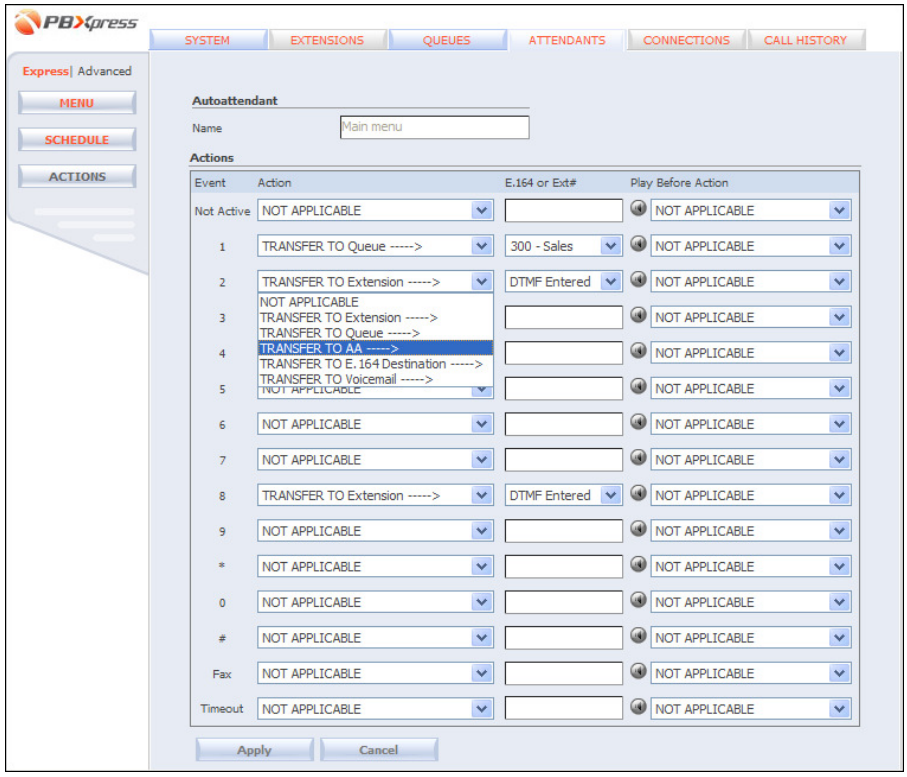
Actions

Here you can define what should happen when the user presses a certain key while in the menu.



Field	Description
Event	• Not Active – The current time is outside the period assigned to this menu. • 1…9, 0, *, # – The user pressed the corresponding key on the keypad. • Fax – incoming fax transmission detected. • Timeout – No response received from the user within 10 seconds.
Action	See the table below.
E.164 or Ext#	Destination for the chosen action, see the table below.
Play Before Action	Play a specific voice prompt before taking action, e.g. play "Your call is being transferred, please wait" before actually forwarding a call to the support team.

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Available actions:

Action	Description
NOT APPLICABLE	This means that the event has no action assigned to it. Play the default prompt and re-play the menu.
TRANSFER TO Extension	Transfer the call to a specific number. Choose one of the extensions, defined in the system (this includes special extension numbers such as voicemail or DISA) or select **DTMF Entered** to allow a user to enter the desired extension number. This action allows you to implement the prompt: "… or if you know your party's extension, please dial it now". If extensions have been created in your system, such as 701, 702, 703, etc., you need only define the action "Transfer to extension entered" for the digit 7 (the first digit in the extension). This also allows you to control which extensions will be directly accessible from the attendant, and which will not. In the example above, you can create a set of extensions for VIPs, such as 601, 602, etc. Other people in the organization can call them on these extensions from their SIP phones, but no one can connect to them directly via the attendant menu.
TRANSFER TO Queue	Chose a specific queue a call will be forwarded to.
TRANSFER TO AA	The IVR will switch to the selected auto-attendant menu.
TRANSFER TO E.164 Destination	Transfer the call to a specific number. Note that this number can be an extension, as well as a number outside your office network.
TRANSFER TO Voicemail	Send a call to the specific voicemail account. In case of a fax event it means that the fax document will be delivered as a new message there.

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Connection Configuration

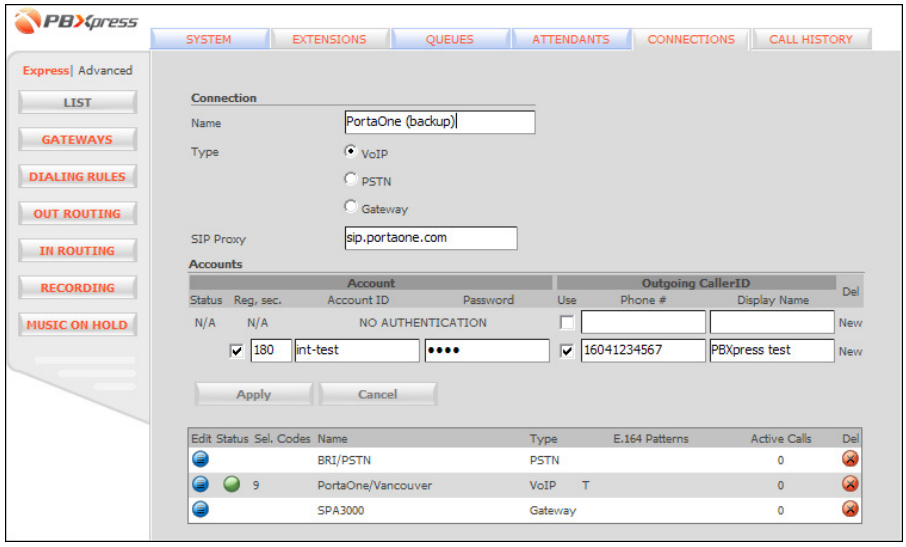
In this section you will learn how to configure your PBXpress system so that it is connected to the rest of the telephony world, and exchanges calls with other carriers.

Create a new connection

When creating a new connection, you must provide the following parameters:

- Name – a unique identifier for the connection.
- VoIP, PSTN or Gateway. If this is a connection to a telco operator or a PBX using one of the PSTN interfaces of PBXpress, choose PSTN. VoIP means that PBXpress will communicate with the ITSP via IP link. Gateway means that a VoIP adapter connected to the PSTN network (e.g. Sipura 3000) on the local network will be used to transmit calls.
- The rest of the parameters depend on connection type (see below).

Creating a new VoIP connection



Available fields:

Field	Description
SIP Proxy	IP address (or domain name) of your provider's SIP proxy.

Using a hostname to define the SIP proxy may potentially create problems in the case of ambiguous domain resolving. PBXpress must not only

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resolve the IP address when sending an outgoing call to a vendor, but also determine which vendor an incoming call arrives from, based on the remote IP address. Thus, connections where the SIP proxy is defined by its hostname can only be used to send outgoing calls to a carrier, but not for matching incoming calls.

Accounts table (you may enter more than one account per connection):

Account information

Field	Description
Reg, sec.	PBXpress should register to the remote SIP server using this account. Without registration, PBXpress may still make outgoing calls, but probably will not be able to receive any incoming calls. If you turn on the registration, you may also specify the re-registration interval.
Account ID	Username for the remote SIP server.
Password	Password for the remote SIP server.

Outgoing Caller ID information

Field	Description
Use	Change the outgoing caller ID. If this checkbox is off, send the same ANI (CLI) phone number and display the name received from the IP phone.
Phone #	Replace CLI (ANI) for outgoing calls with the specified value.
Display name	Replace caller info for outgoing calls with the specified value.

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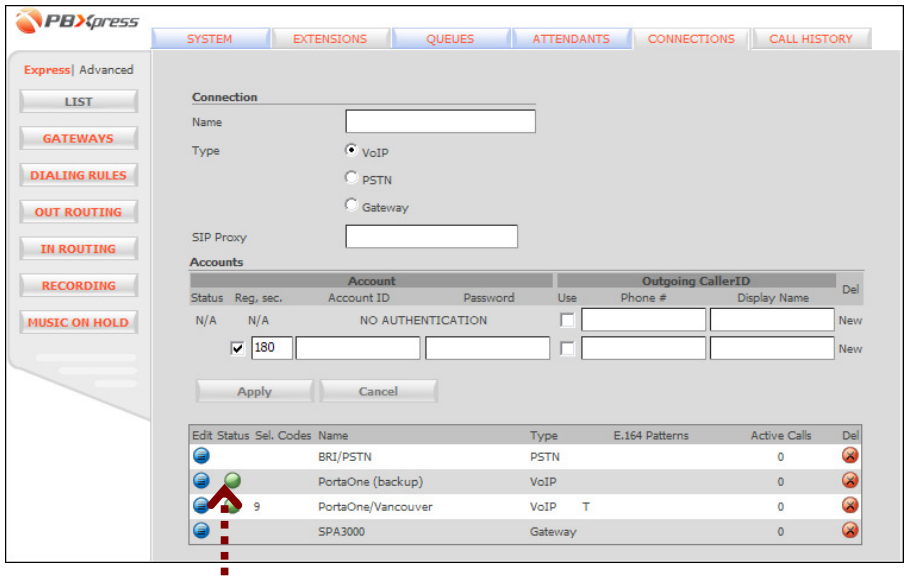
If this connection assumes registration to a remote SIP proxy, you will see a green light in the **Status** column after the connection is successfully configured. If some accounts are not registered to the SIP server, while others are, a blue light will be shown.



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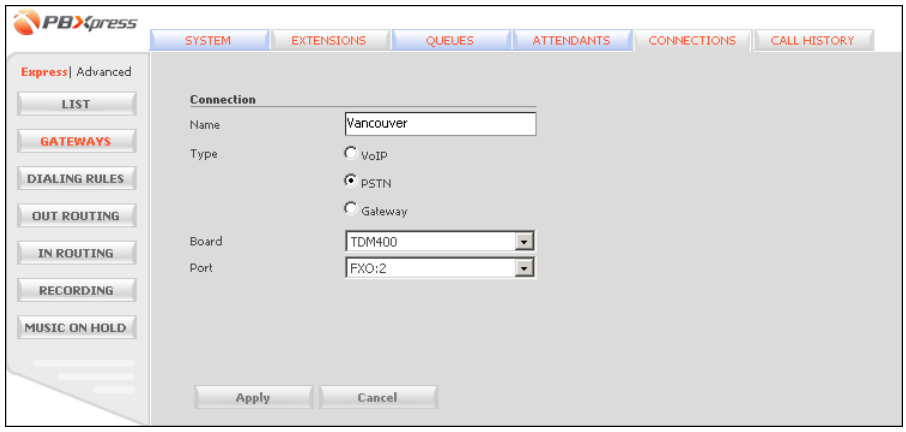
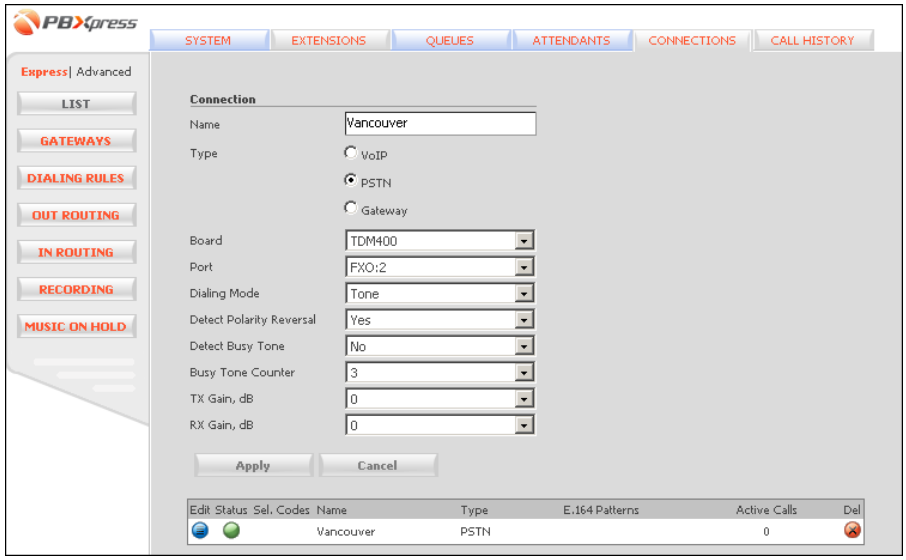
Creating a new PSTN connection (FXO)



Available fields:

Board	ID of the PSTN interface card installed in the system. This is required in order to distinguish between several PSTN cards installed in PBXpress.
Port	A specific port on the card (for instance, a single card may have two FXO ports).

Some of the connection parameters may not be available when creating a new connection, and can only be accessed when changing existing connection information.



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Available parameters:

Field	Description
Dialing Mode	On analog lines you have the option of tone or pulse dialing (pulse dialing is used in some older telephony systems). Choose the option applicable to your phone line.
Detect Polarity Reversal	Polarity reversal is required to provide correct answer supervision (to detect when a call is actually connected) on analog lines. This feature requires support from both the phone company and your PSTN equipment, so activate it only after you have confirmed that this is supported by your telco. Being able to detect call connect time correctly is required for two purposes: displaying the correct time in CDRs, and playing a "this call is being recorded" announcement for the called party on time. Otherwise, the announcement will start playing immediately after the number is dialed, and will probably finish playing before the called party picks up the phone.
TX Gain, dB	Increasing TX Gain will boost the voice volume of the outgoing conversation.
RX Gain, dB	Increasing RX Gain will boost the voice volume of the incoming conversation.

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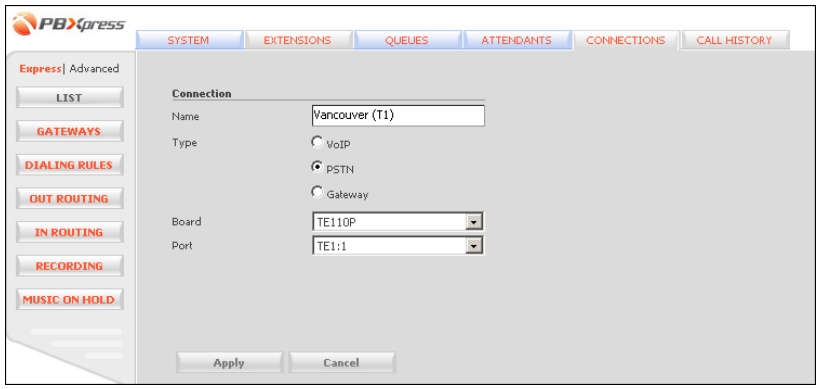
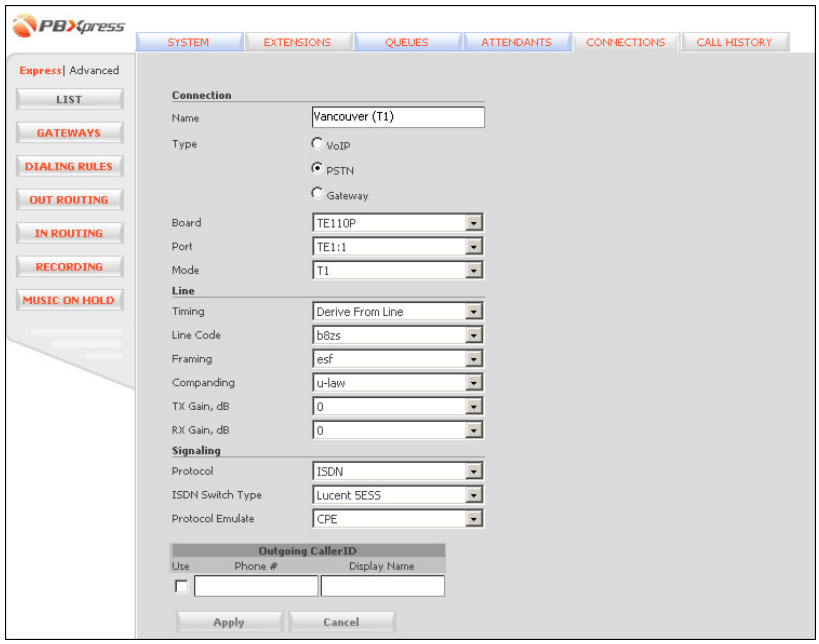
Creating a new PSTN connection (E1/T1)



Available fields:

Board	ID of the PSTN interface card installed in the system. This is required in order to distinguish between several PSTN cards installed in PBXpress.
Port	A specific port on the card (for instance, a single card may support quad T1/E1 interface).

Some of the connection parameters may not be available when creating a new connection, and can only be accessed when changing existing connection information.



Most of the parameters are dependent on the type of PSTN connection; please obtain the required information from your telco carrier.

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Line parameters:

Field	Description
Timing	Possible values are **Provide To Line** (PBXpress provides clocking), **Derive From Line Primary** (PBXpress receives clocking from the line), and **Derive From Line Secondary** (PBXpress receives clocking from this line if the primary line is not available).
Line Code	For T1 ports, possible values are **ami** and **b8zs**. For E1 ports, possible values are **ami** and **hdb3**.
Framing	For T1 ports, supported framing options are **d4(sf)**, **esf**. Supported framing options for E1 ports are **ccs**, **ccs-crc4**, **cas**, **cas-crc4**.
Companding	The available options are u-law and a-law.
TX Gain, dB	Increasing TX Gain will boost the voice volume of the outgoing conversation.
RX Gain, dB	Increasing RX Gain will boost the voice volume of the incoming conversation.

Signaling parameters:

Field	Description
Protocol	**ISDN** or **CAS** (channel-associated signaling).
For ISDN signaling protocol	
ISDN Switch Type	The available options are **AT&T 4ESS**, **EuroISDN**, **Lucent 5ESS**, **National ISDN type 1**, **National ISDN type 2**, **Nortel DMS100.**
Protocol Emulate	If the device is to act as an end-point (**CPE**) or emulate the ISDN network side (**Network**). If you are interconnecting two devices, e.g. PBXpress and a legacy PBX, one of them must be configured as CPE and the other as network. Please see ***Error! Reference source not found.*** for instructions on how to create the physical cable for such an interconnection.
For CAS signaling protocol	
Type	The currently supported signaling types are: FXS Ground Start, FXS Loop Start, FXS Kewl Start, FXO Ground Start, FXO Loop Start, FXO Kewl Start, E&M, E&M Wink - DNIS - DTMF, Feature Group D - DTMF, Feature Group D - MF, Feature Group B - MF, SF, SF Wink - DNIS - DTMF, SF Feature Group D - DTMF, SF Feature Group D - MF, SF Feature Group B - MF. Few notes: • Kewl Start supports disconnect supervision via line drop and battery reversal. • Feature group D supports the transfer of ANI and DNIS information.

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Outgoing Caller ID information (only available when DNIS information is supported by the currently selected protocol, e.g. ISDN signaling or CAS Feature Group D):

Field	Description
Use	Change the outgoing caller ID. If this checkbox is off, send the same ANI (CLI) phone number/display name received from the IP phone.
Phone #	Replace CLI (ANI) for outgoing calls with the specified value.
Display Name	Replace caller info for outgoing calls with the specified value.

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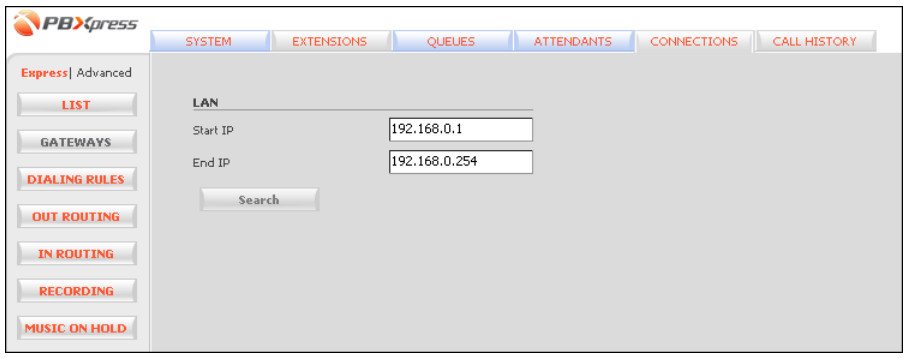
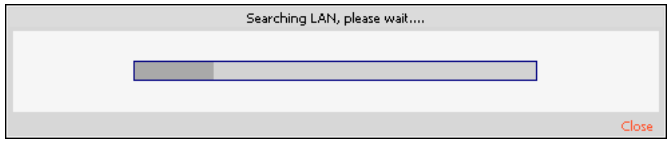
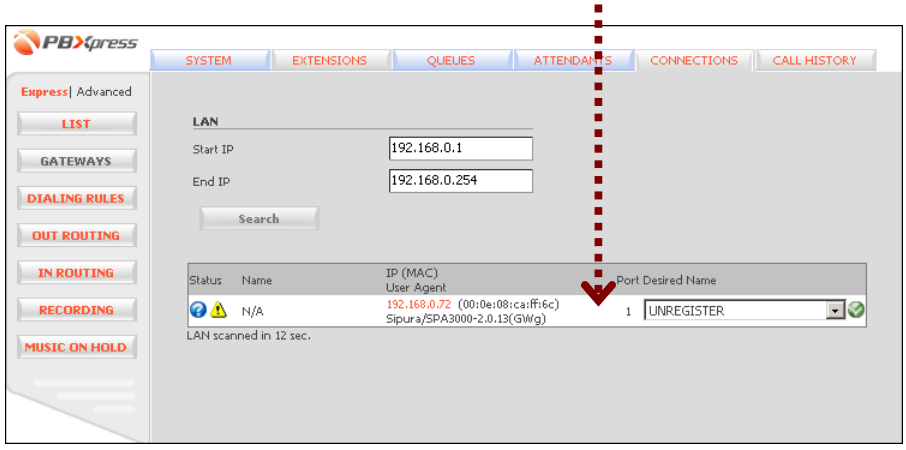
Gateways

PSTN-VoIP gateways on your LAN can be automatically detected and provisioned for use in transporting calls. Before you proceed with the steps described below, make sure that you have already created a connection of the **Gateway** type.



Specify the starting and ending IP (in most cases, the default values provided are OK) and press **Search**.





You will see a list of the VoIP gateways which were detected on your internal network.

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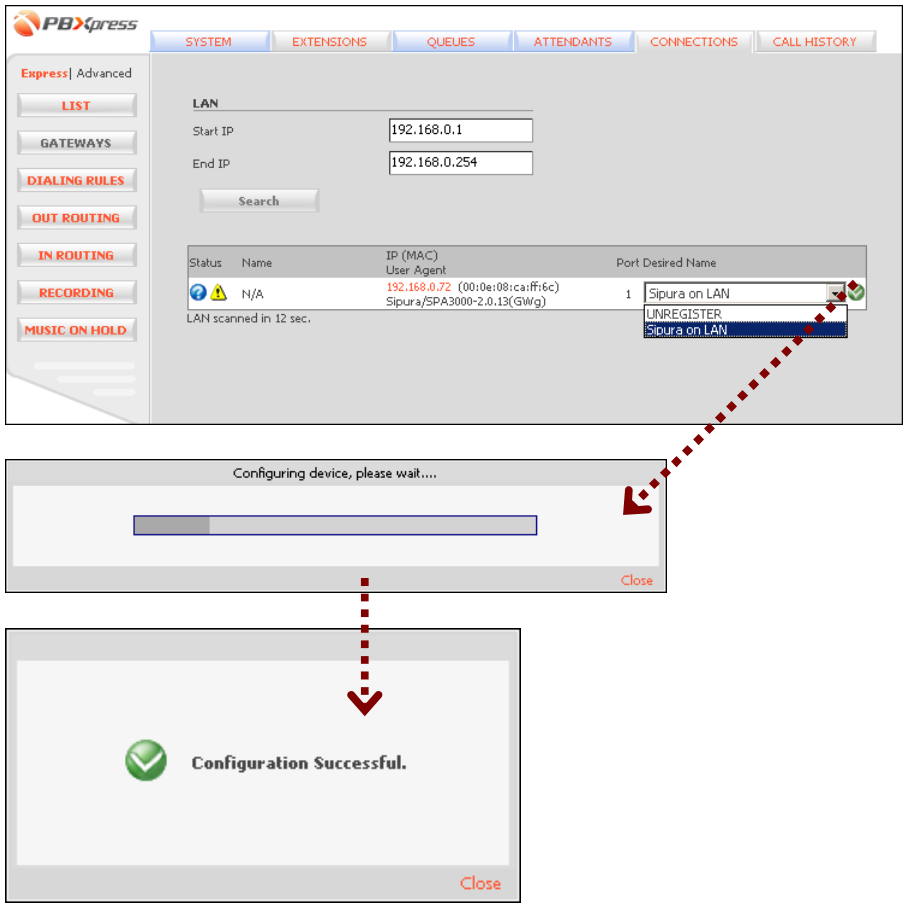
Choose one of the available connections from the select menu, and then click on the button to auto-configure the device.



Press **Close** to close the pop-up window and return to the gateway configuration screen. PBXpress will re-scan the network and display the current status of the VoIP gateways.

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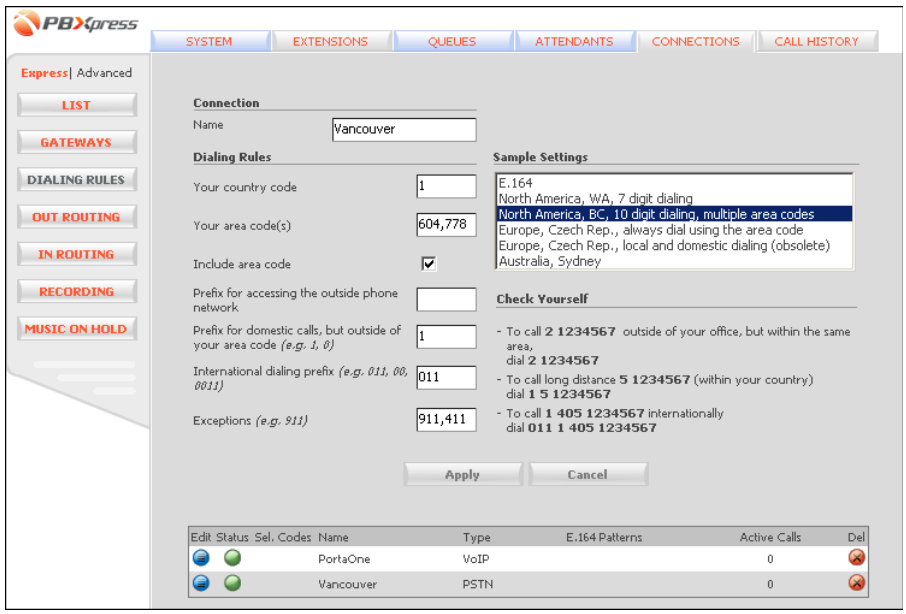
Dialing Rules

In most cases, a carrier will require a specific numbering format that does not match the one you use in your telephony network. For instance, if you are based in the USA and place a call to the local telco, you will need to dial the 011 prefix, even if your users never dial it on their IP phones. Thus, PBXpress must convert the number from the format used in your system to the carrier-specific format. This is done by specifying dialing rules, similar to the ones we have configured for our users.
Double-click on the corresponding row in the **Sample Settings** list box to load the sample settings.



Field	Description
Your country code	The E.164 country prefix, e.g. 1 for USA & Canada, or 420 for the Czech Republic.
Your area code(s)	The code for the area or city your phone belongs to. If there is more than one area code considered to be local (for instance, area codes 604 and 778 in Vancouver, BC), enter all of them, separated by commas.
Include area code	If checked, this means you always dial the number including area code, even when dialing within your "local" area. For instance, even when dialing a local number within the 604 area code, you still must dial 604 NNN-NNNN.
Prefix for accessing the outside phone	Dial (if applicable) to get an outside line (e.g. when using some office PBX systems, you first must dial 0 and then the actual number).

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network	
Prefix for domestic calls	Dial before the phone number when dialing domestic long-distance (e.g. 1 in the USA & Canada, 0 in many European countries).
International dialing prefix	Dial before the phone number when making an international call (e.g. 011 in the USA & Canada, 00 in many European countries).
Exceptions	Comma-separated list of special numbers. These numbers will not be treated according to dialing rules, but rather processed "as-is". Usually these are emergency numbers (911 in the USA, 112 in Europe).

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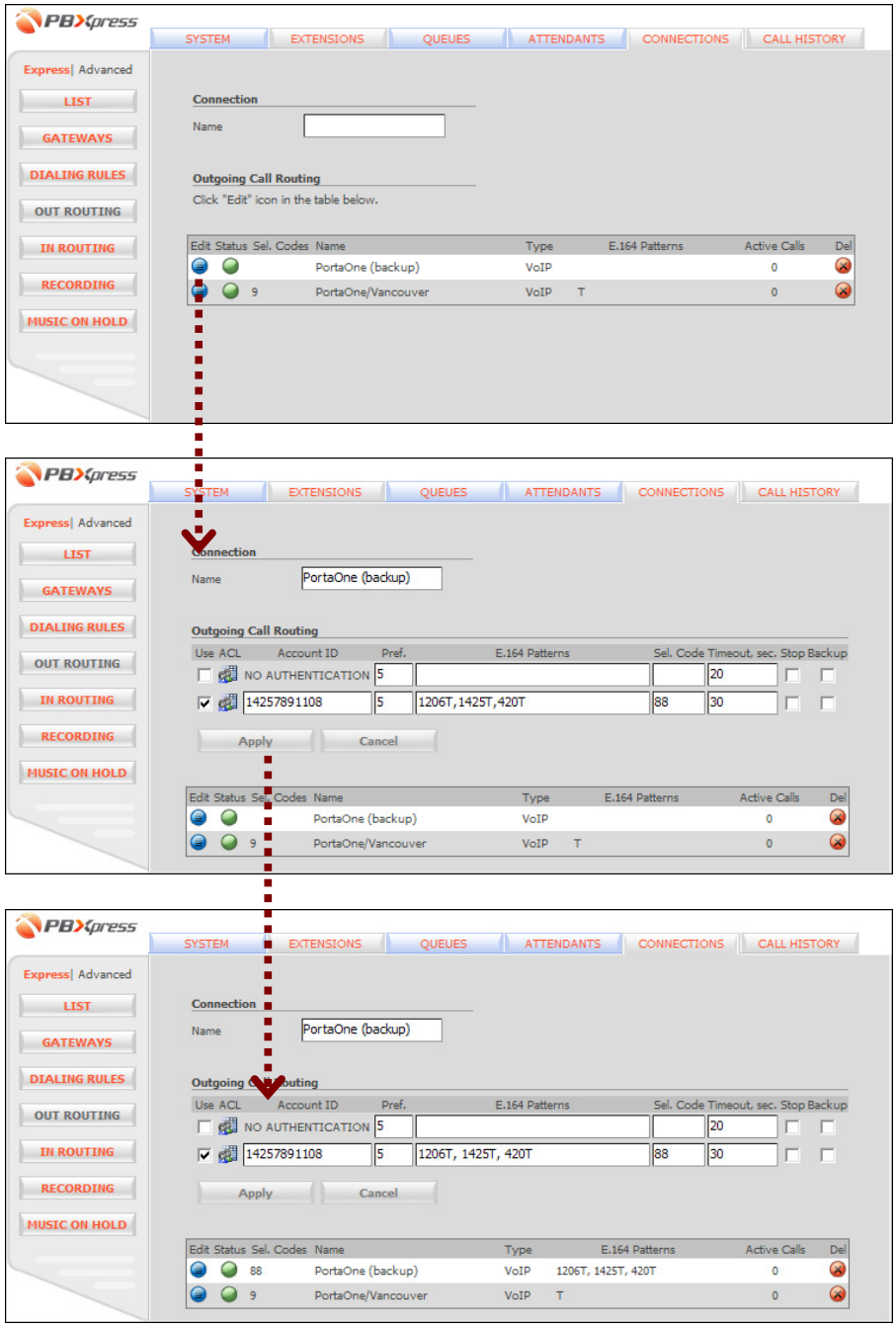
Out Routing

On this screen you can configure which types of outgoing calls will be sent via this connection.



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Field	Description
Use	Include this connection (or a particular account, if multiple accounts are defined for the connection) in the outgoing call routing.
ACL	Click on the icon to edit the list of extension classes, which are allowed to place outgoing calls via this connection (see below).
Account ID	This read-only field displays a specific account, which you apply the settings to. PBXpress allows you to have several accounts associated with the same connection. For instance, for better cost tracking you may use account A for all extensions in the sales department, while account B will be used for all other extensions – thus, all calls will be clearly separated on a bill from your vendor.
Preference	Enter a positive number (1-99). This allows you to manually adjust the routing order where several carriers terminate the same prefixes.
E.164 Patterns	A comma-separated list of prefixes which are to be routed via this connection. Each prefix must end with T (wildcard), otherwise it will be considered a specific number. See the following examples: • "1604T" means that calls to every number starting with 1604 (e.g. 1604-123-4567) will be routed through this connection. • "911" means that only calls made to the specific number 911 will be routed via this connection. If you wish to use a connection to route a call to 91118765432 (Delhi, India), specify "911T" among the prefixes for this connection. It is possible to use . (dot) as a wildcard, meaning "one single digit". This is useful when routing only numbers of a specific length, e.g. extension numbers. Thus the pattern "3.." will route the numbers 301 or 380, but not 380442207615. **NOTE:** Star (*) is no longer used as a wildcard symbol, and now simply denotes the star key, as dialed from the phone's keypad.
Sel. Code	A special code, which may be dialed before the destination number to override normal call routing and utilize this particular connection.
Timeout, sec.	If a call is not connected within the specific interval, use the backup connection.
Stop	Apply the "hunt stop" to this connection – if PBXpress tried to establish an outgoing call via this connection and the attempt failed – do not try any other connections (although some may be available).

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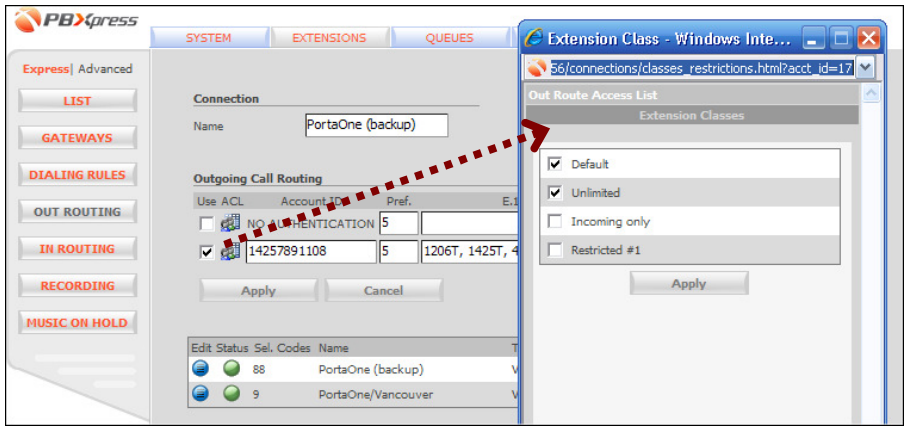
Backup	Use this connection as a backup, in case other connections fail, even if the phone number prefix has not been specified for it.

ACLs for out routing

You may specify which extension classes are allowed to use this connection.



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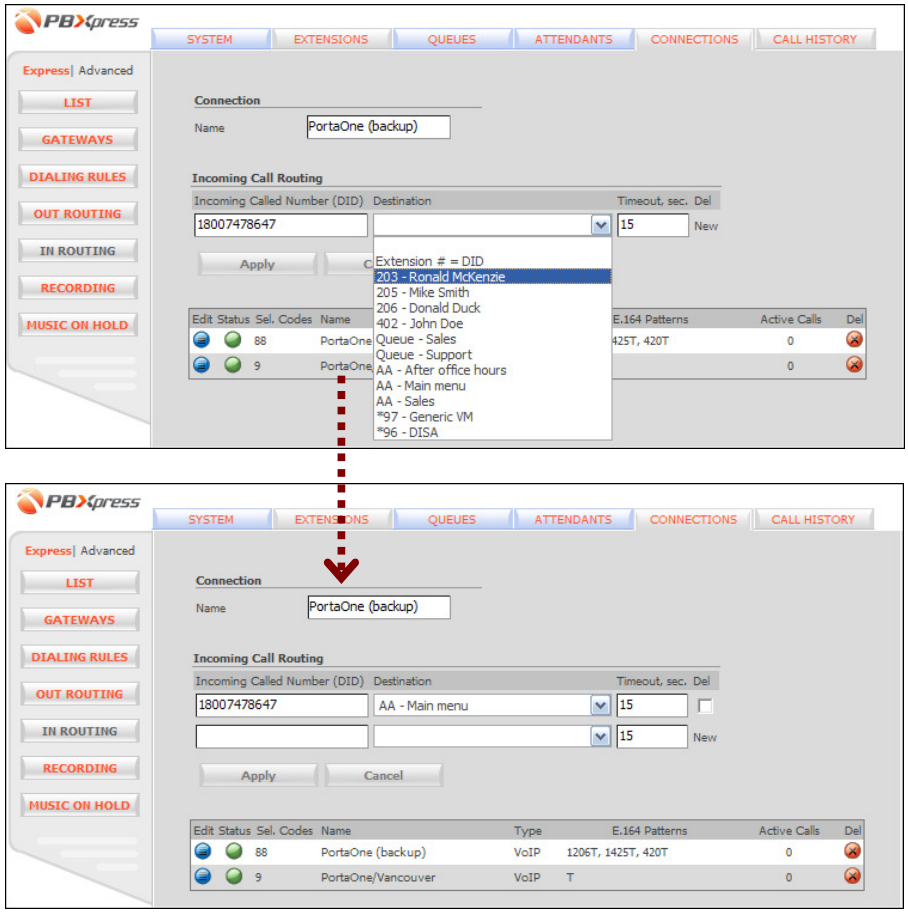
In Routing

"In routing" defines where an incoming call arriving via this connection will be delivered to.
Note that the call may be forwarded to an individual extension, as well as to a queue or an attendant.
It is quite possible for a single connection to deliver calls to multiple phone numbers. For instance, your T1 connection can deliver incoming calls to several office numbers, as well as your toll-free number. In this case, you need to differentiate between such calls, forwarding each of them to the proper extension or queue; this is called "direct inward dialing" (DID). The protocol used on the phone line must support the transfer of DNIS information, which can be accomplished if either ISDN or CAS Feature Group D signaling is used or if your connection is of VoIP type.



Choose the extension and press **Apply**.

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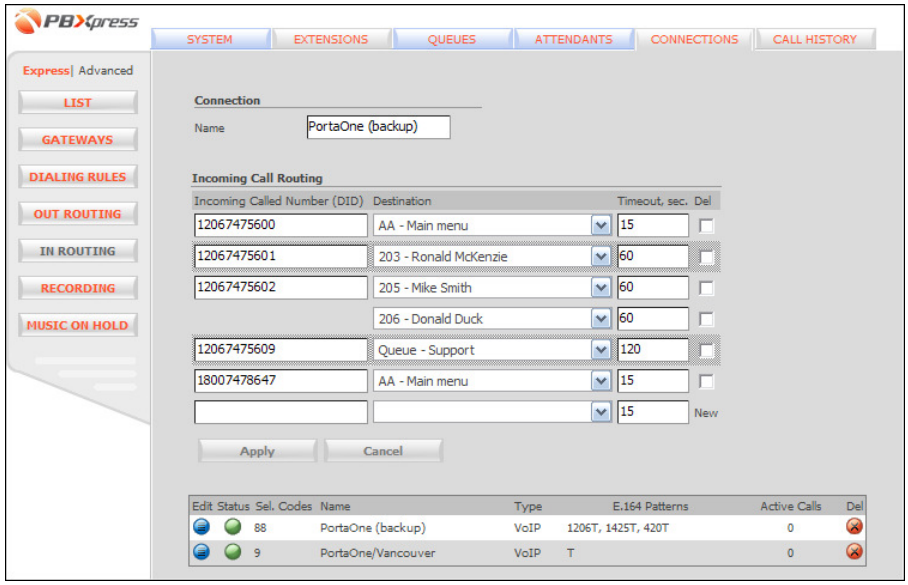

You may forward the incoming call to several extensions or queues. PBXpress will try to ring all of them in the order specified in the configuration. In the example above an incoming call to 12067475602 will first ring on Mike's phone and if not answered there it will start ringing on Donald's phone.

Hint: When defining call forwarding to a queue with several extensions and a ringing policy other than **ringall** (so that the extensions will be tried sequentially, and only one extension will ring at a time), make sure you specify a large enough timeout to ring all of the extensions.

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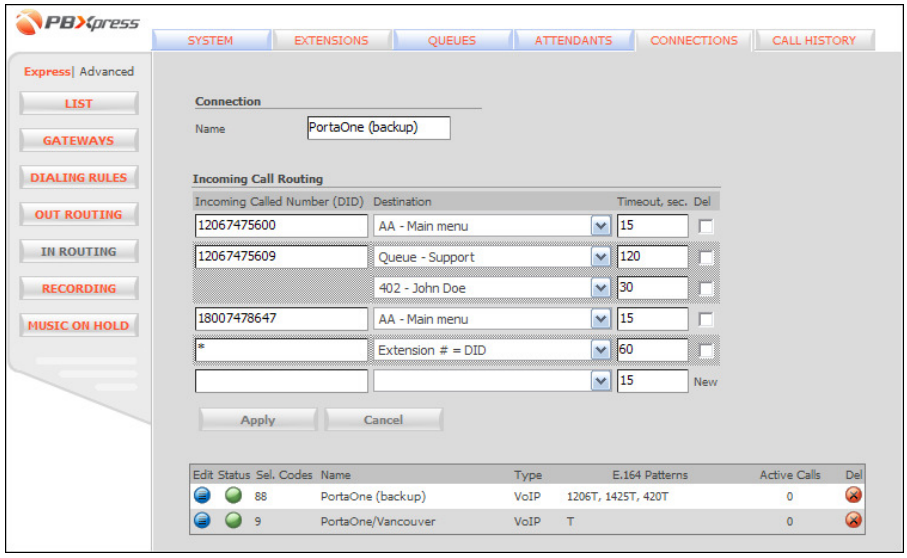
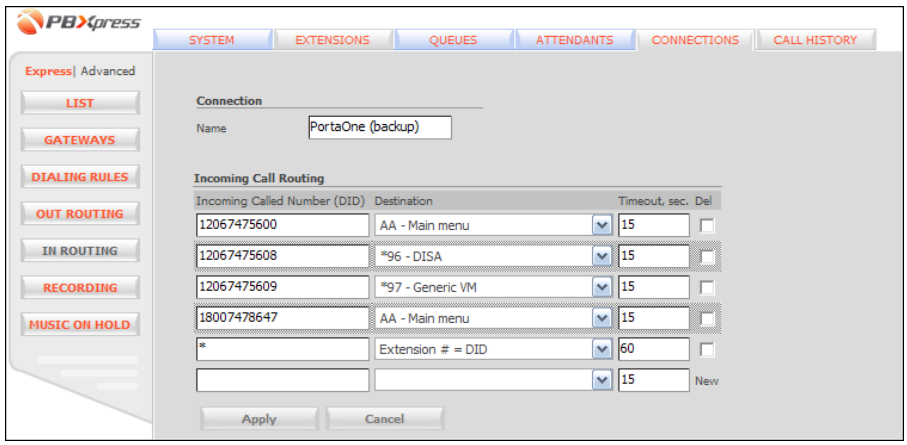
In routing with multiple incoming numbers (DID)

Typically incoming calls are delivered to the corresponding extension. PBXpress can automatically match the DNIS number received from the remote carrier to the PBXpress number (this includes both extension and the full phone number associated with an extension). To accomplish this, out * (star) into **DNIS** and choose "Extension # = DID" in the **Destination** select menu as shown below:



You can forward an incoming call to a certain number or to a specific extension. For details see the configuration for numbers 12067475600, 12067475609 or 18007478647 above.
Other possible options for incoming call redirection are DISA or generic voicemail. There the user will be prompted to enter his extension number and password to get access to the corresponding service.



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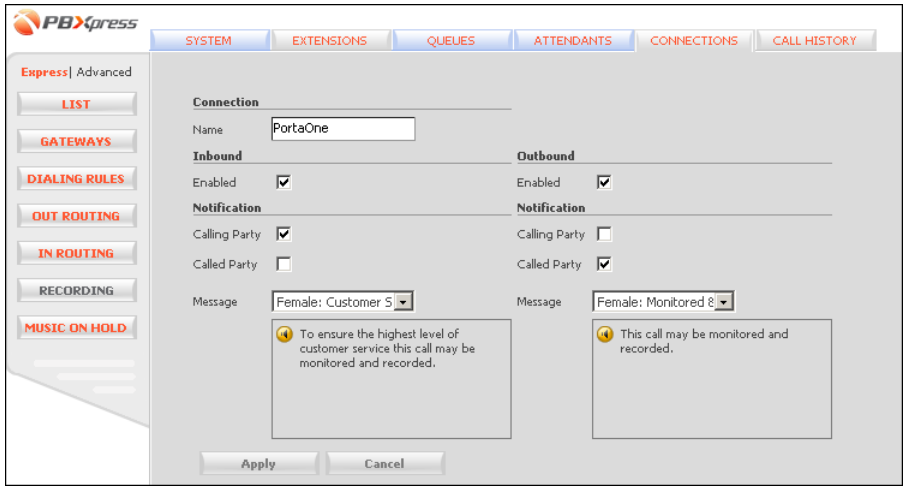
Recording

The Recording configuration screen for a connection is basically identical to the one already seen in the Extensions section. By setting up the parameters here, you can define whether calls going through this connection should be recorded. There are independent configuration options for incoming and outgoing calls.



Field	Description
Enabled	Turn call recording on or off
Calling Party	Notify the caller that the call will be recorded before the conversation starts.
Called Party	Notify the recipient of the call.
Message	Which voice prompt should be used for the call recording announcement.

In case if a call goes from node A to node B (both nodes are the part of the same cluster), and is terminated via a connection on node B – notification is done according to the policy of the originating node (A).

NOTE: Only voice prompts of the **Notification** type will be accessible in the select menu.

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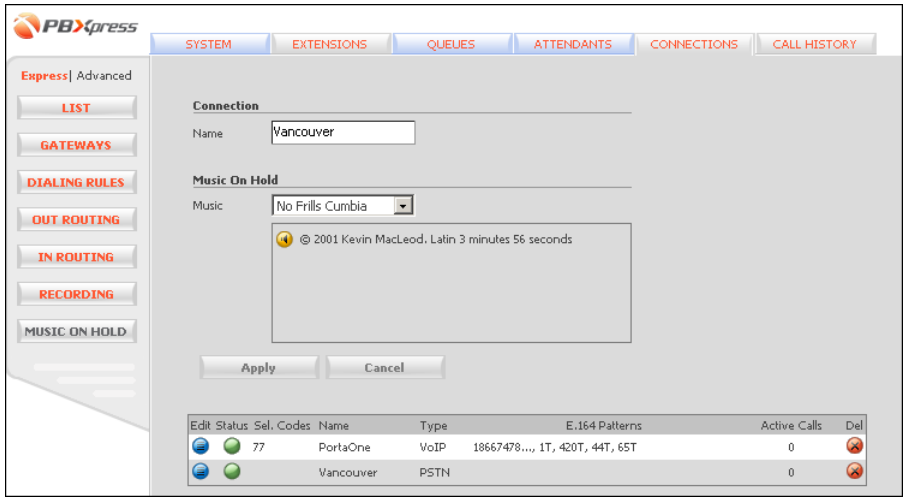
Music on Hold

Again, this screen is very similar to the one in the Extensions or Queues section.



For each connection you may choose which music file is to be played when the user is put on hold.

NOTE: Only voice prompts of the **MOH** type will be accessible in the select menu.

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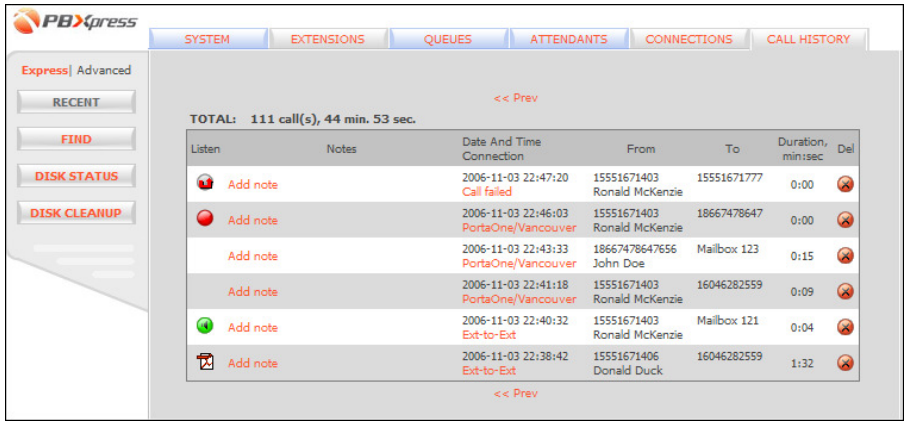
Using the System

After you have completed the steps described in the *System* section, you are ready to make calls. There are only a few administrative tasks you will be responsible for.

Monitoring Call Records

For every call made through PBXpress, an entry is made in call history. You will be able to see when the call was made, from which phone number (extension) it was originated, what the destination number was, and other information. There are two options in the main **Call History** tab: **Recent** allows you to quickly see the most recent calls made, while **Find** allows you to perform an advanced search to display specific call records.

Recent



Here all the calls registered in the system are displayed in reverse chronological order, so that the most recent calls are shown on the first page. For your convenience you can see the total number and duration of calls on top of the call list. You can always go back in history by moving to the next page.
You can also listen to recorded call conversations. The meaning of the icons in the **Listen** column is as follows:

- - The call was recorded, and the recorded file is available in the PBXpress internal storage, so you can listen to the record by clicking on the icon.
- - The call was recorded, but the recorded file is no longer available in the internal storage, since it has been moved to a

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backup optical disk. In order to listen to the file, click on the icon. You will see a dialog box similar to the one below:

Locate the optical backup disk with the matching label, insert it into the CD/DVD drive, and press **OK**.

- - The call was recorded, and the recorded file is available on the backup optical disk which is currently in the drive. Click on the icon in order to listen to the call record.
- - The call could not be connected.
- - The call has been blocked (e.g. extension class does not allow to place calls to the dialed destination).
- Empty icon – A call record is not available. This means that the call was unsuccessful, or was not recorded because call recording had been switched off for this connection at that time.

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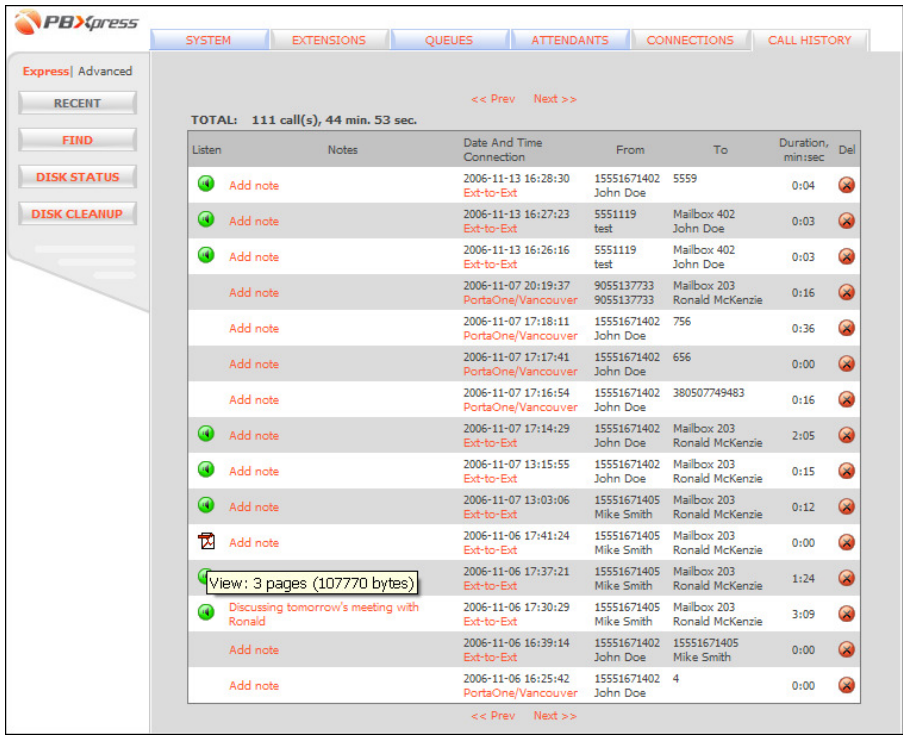
Fax messages

Fax is stored in the voice mailbox as a special type of message. Instead of listening to it – you can click on the **Fax** icon and display the fax in your web browser. For fax calls, the total number of pages in the document is displayed if you move the mouse cursor over the **Fax** icon.



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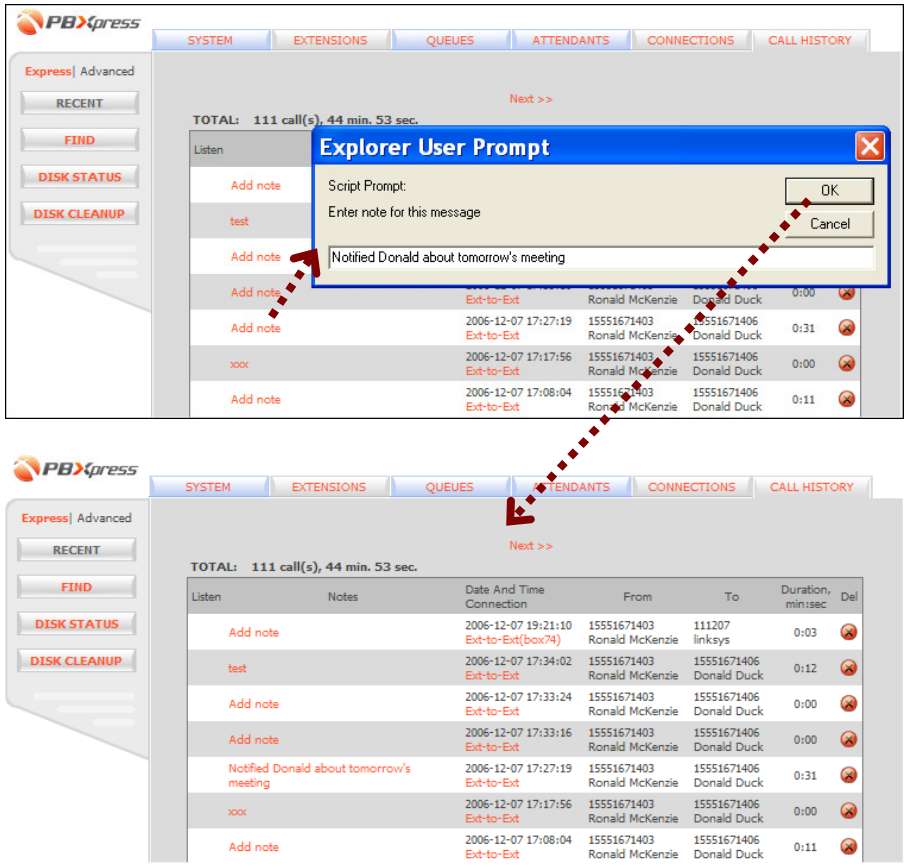
Call notes

Each call has a Notes field, where you can enter your own identification string for this call. Click on **Add note** to add a call description, or click on the note text to edit the existing note.



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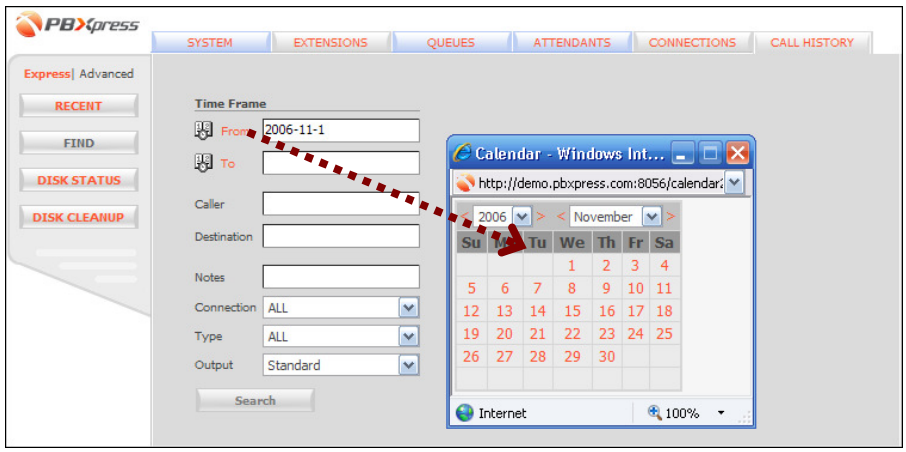
Find specific calls

You can search for calls in a certain date range. Moreover, you can apply additional filters to the call search.

Field	Description
From	The report period start date (YYYY-MM-DD). You can click on the icon to summon a calendar pop-up.
To	The report period end date (YYYY-MM-DD). You can click on the icon to summon a calendar pop-up.
Caller	Call originator (enter the full phone number instead of the extension number).
Calling Party	Destination phone number (for extension-to-extension calls you still should enter the full phone number instead of the extension number).
Notes	Text in the note field.
Connection	Only search among calls which went through a specific connection.
Type	Display all calls or successful (connected) calls, failed calls or fax calls only.
Output	The desired format of the CDR list • Standard – display results as a web page (HTML) in your browser • CSV – download CDRs as a .CSV (comma-separated value) file • XLS - download CDRs as an .XLS (Microsoft Excel) file



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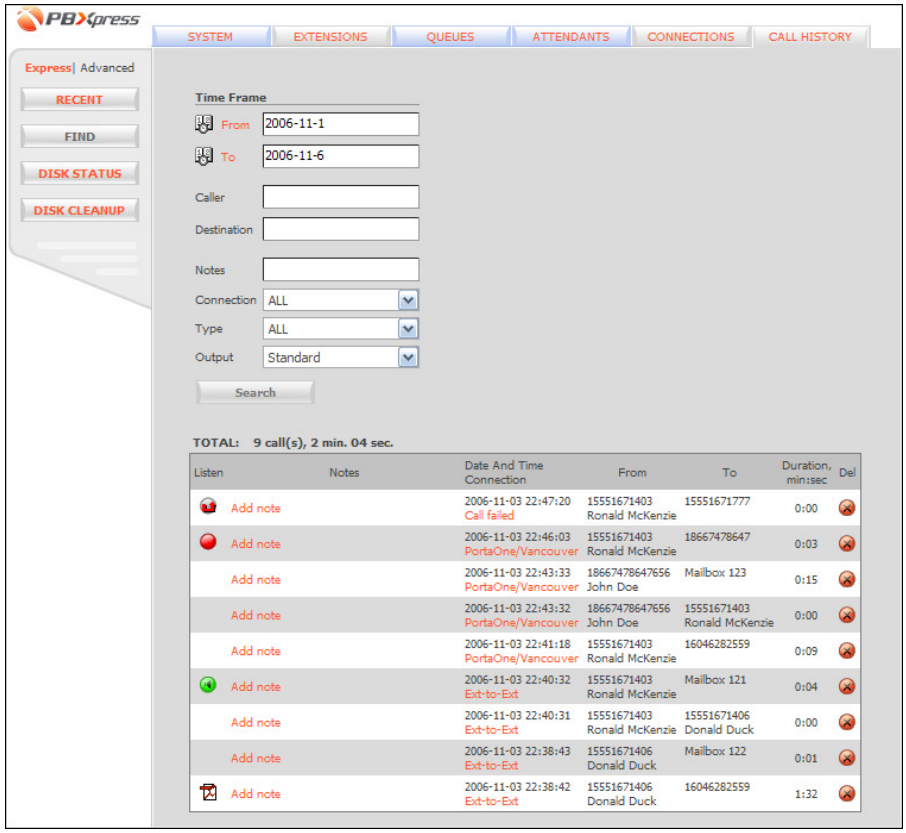
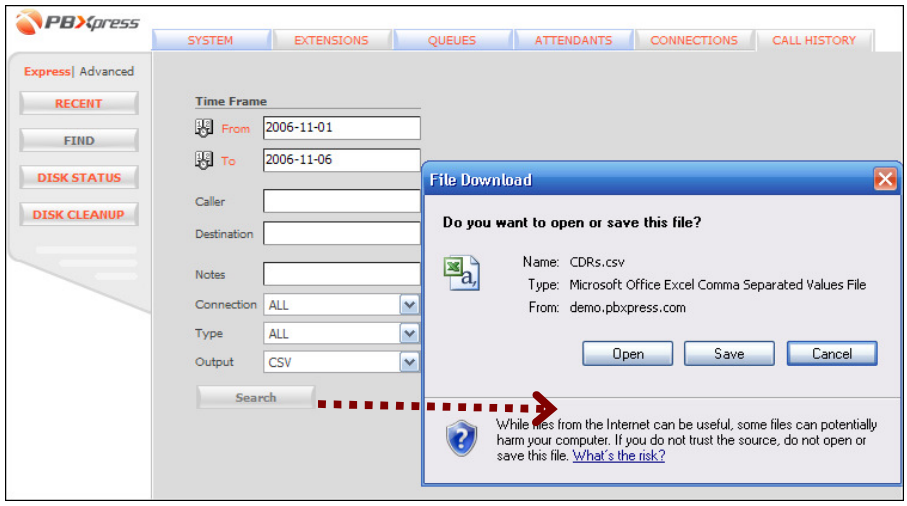


Below you may find an example of downloading call history in CSV format.



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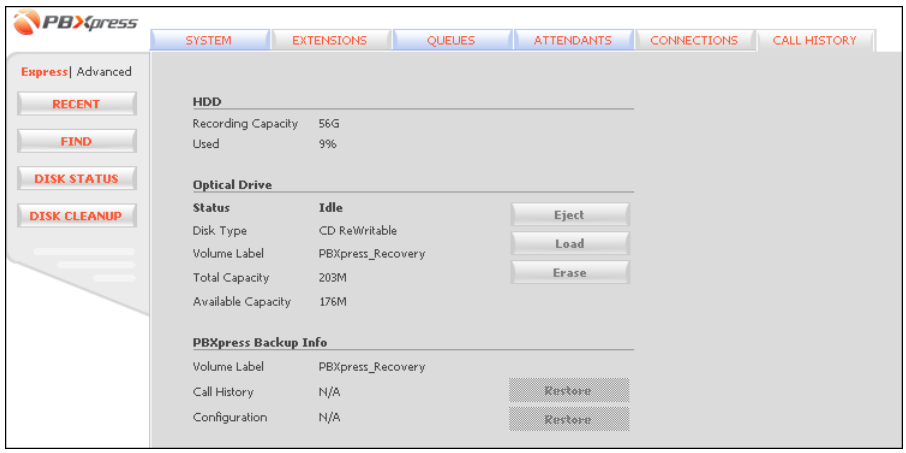
Moving Old Call Records to Backup Storage

PBXpress' internal storage can hold a sufficient amount of recorded conversations, but it is not unlimited. This is why it is recommended that you periodically copy old call records to optical disks, thus clearing internal storage space for new calls.

NOTE: When you move call records to an optical disk, only the sound file associated with the call is erased from the internal storage. The call record with information such as call time, origination/destination number, and the like is still available.

Status

Choose the **Call History** tab, and then click on the **Disk Status** button.



HDD section

This is for information purposes only. You can see how much space is available in total for voice recording, and how much is currently being used.

Optical drive section

Here you can see information about the disk currently in the drive, eject it from the drive, or load it.

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PBXpress backup info

If the optical disk in the drive contains backup information, a brief summary is shown here.

Field	Description
Volume Label	Each backup disk is uniquely labeled (a disk ID is assigned when the disk is written).
Call History	This field contains a from-to date of the call set written on this disk. It is possible to copy the call records back to the internal storage using the **Restore** button. Note that this is not required in order to listen to a voice conversation. Rather, it is sufficient to place the optical disk in the drive.
Configuration	In addition to call records, it is also possible to store the current PBXpress configuration on an optical disk. This will include configuration settings such as extensions, connections, and so on. Thus, if your configuration becomes damaged due to improper action by the administrator, or if you performed a factory reset on your PBXpress, you may quickly restore your previously stored configuration.

Make sure you periodically perform a backup of your PBXpress configuration. This will allow you to quickly restore the system to its operational state if there is a problem (e.g. hardware failure) which requires system reinstall.

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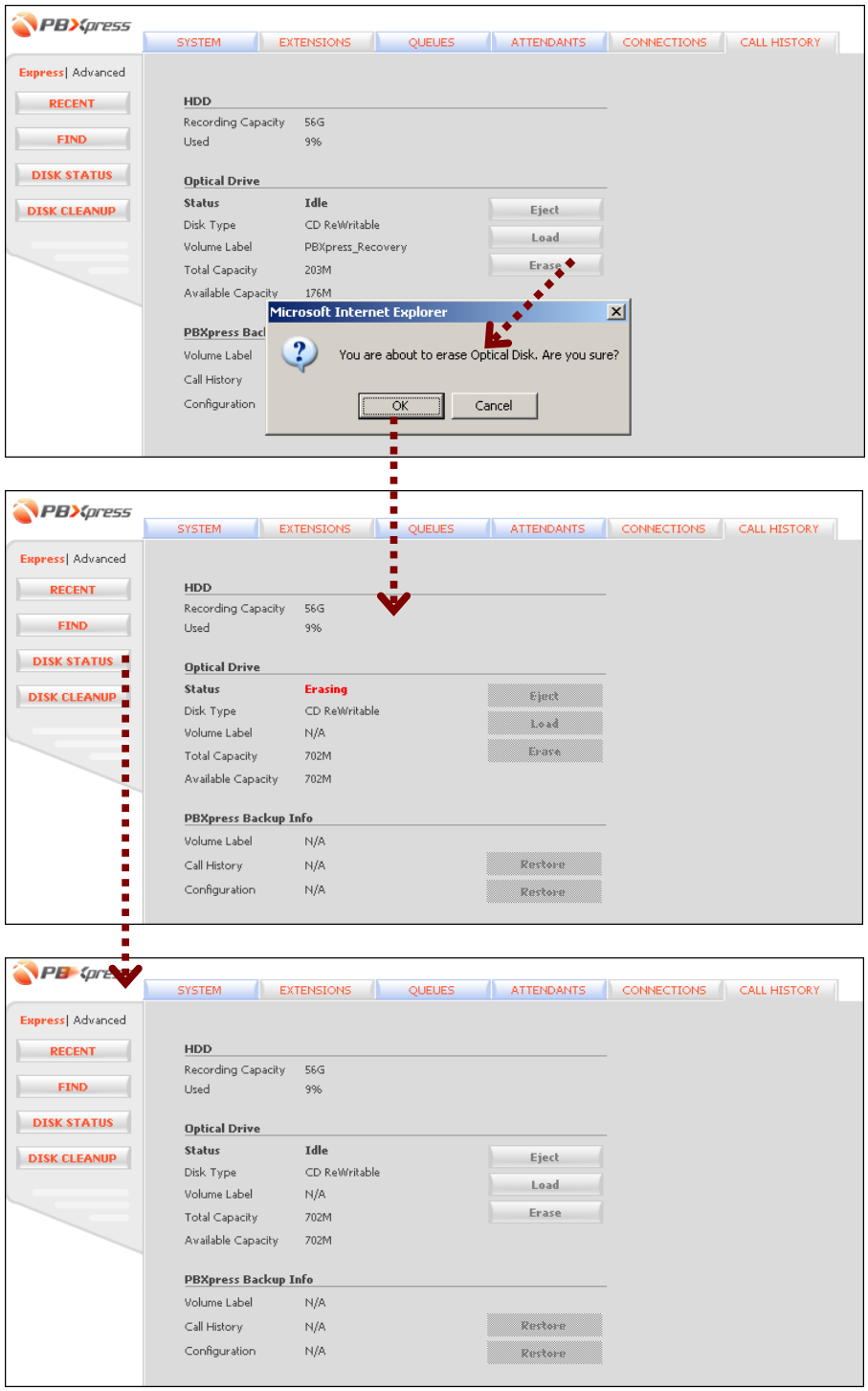
Erasing a re-writable optical disk

If a disk is re-writable, you can also erase it.



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Recording a Disk

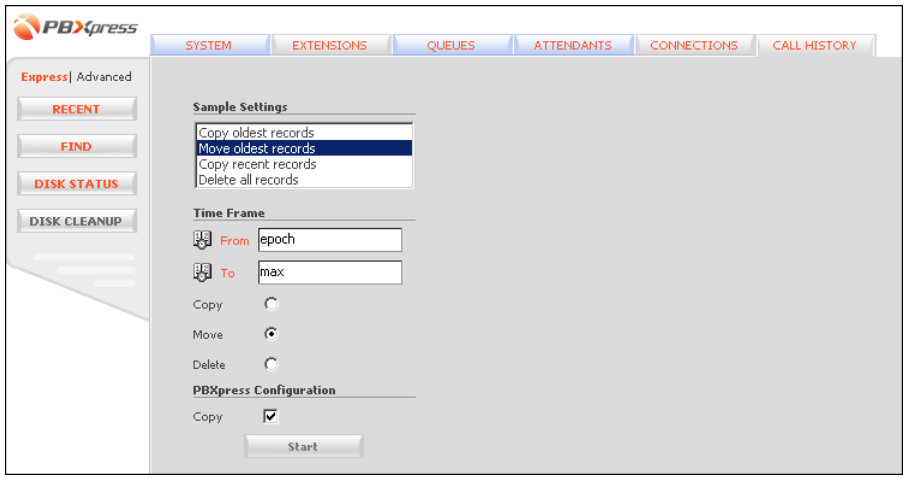
Press the **Disk cleanup** button on the left to access this screen.

You can specify the date range for the calls to be placed on the disk. There are also two options: Copy and Move.

- There are two special keywords you can use as a value for the date field. **Epoch** means "the earliest date possible", and **max** means "the latest date possible".
- **Copy** means that the sound files will not be deleted from the PBXpress internal storage, so there will be no change in the available disk space. This option is convenient if you need to make a special copy of calls within a certain period (e.g. to do an evaluation of calls made during a training session).
- **Move** means that the sound files will be deleted from the internal storage after being copied to the disk. This is the normal routine.
- **Delete** means that the sound files should just be removed, not copied anywhere.

For your convenience, some sample settings are provided:

- Copy oldest records – Starting with the earliest date possible, this copies the call records onto the optical disk (as many as will fit there). The original sound files will stay on the internal storage.
- Move oldest records – Same as the above, except the sound files will be deleted after the optical disk is written. This allows old call records to be backed up periodically, and frees up space for new calls.
- Copy recent records – Allows you to make a copy of all calls which have not been backed up yet.
- Delete all records – Remove the old entries.



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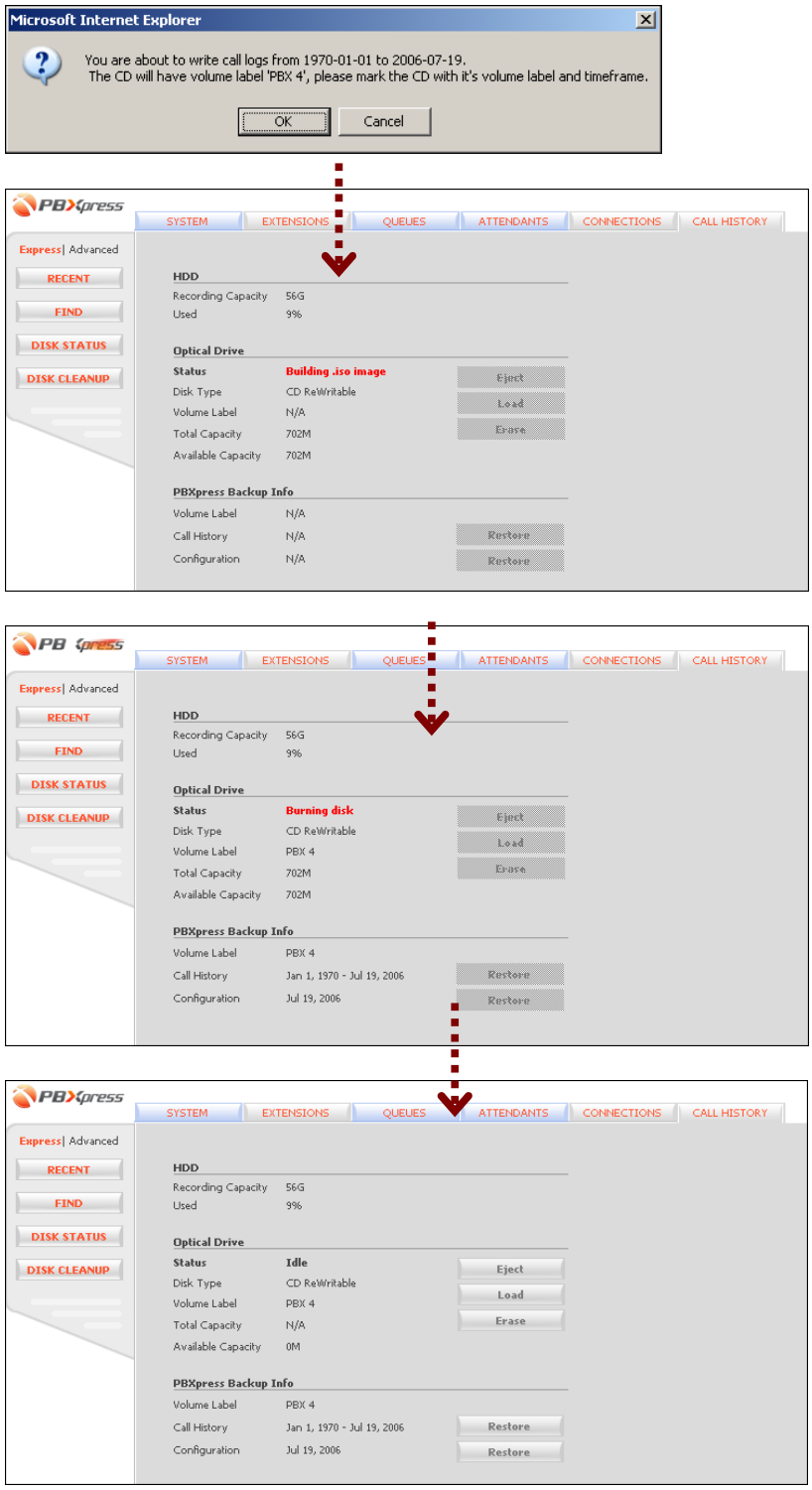
Press **Start** to begin the recording process.
The following warning message will appear ; press **Ok** to continue.



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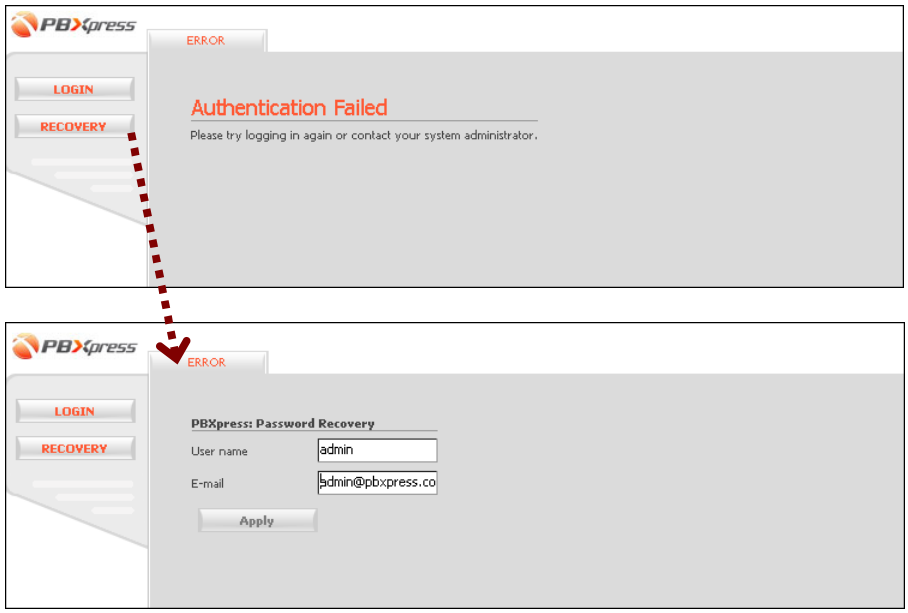
Recovering a Lost Administrator Password

If, for some reason, the password for the administrator's web access is no longer available, you can still restore it. After three unsuccessful logins to the web interface, you will see an Authentication Failed screen similar to the one below. After this, click on the **Recovery** button and enter a valid administrator username and email address. PBXpress will generate a new password and send it to the specified email address.



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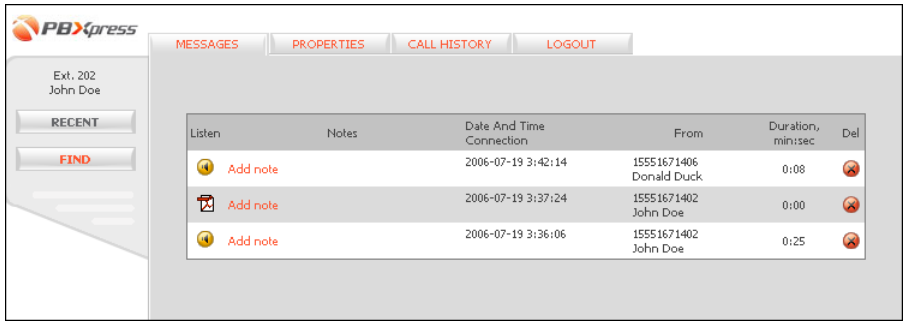
Extension Self-care Interface

This is a separate web interface and, in contrast to the PBXpress administrator interface, it is accessible by end users. To begin, open your web browser and type in the voicemail web interface URL: **http://N.N.N.N:8000**, where N.N.N.N is the IP address or domain name of your PBXpress box.
You will be prompted to enter a username (your extension number) and a password (assigned to you by the administrator; by default, the password is identical to your extension number).

Messages



After you log in, you will see the recent messages screen. You can:

- Listen to the messages (click on the **Listen** icon for the corresponding message). The icon indicates that the message is old (has been played already).
- View faxes (click on the **Fax** icon for the corresponding message)
- Enter your own comments for the voice message (click on **Add note** to add a new text description, or click on the note text to edit it).
- Erase voice messages (click on the **Delete** icon next to the message).

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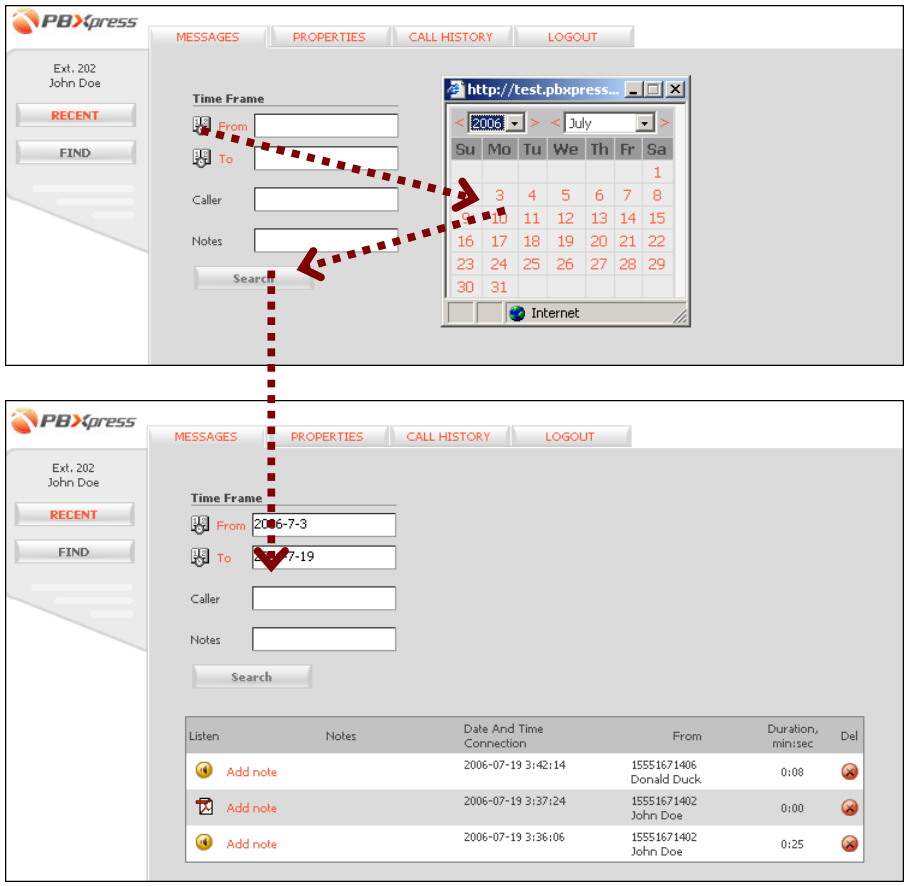
Using the **Find** screen, you may display messages as follows:

- For a specific time interval (you can either enter dates as year-month-day, or use a pop-up calendar).
- Only from a specific caller or callers (you do not need to enter the full phone number; as shown below, if you specify just 1866, all calls from phone numbers beginning with 1866 will be displayed).
- Containing a certain text string in the note.



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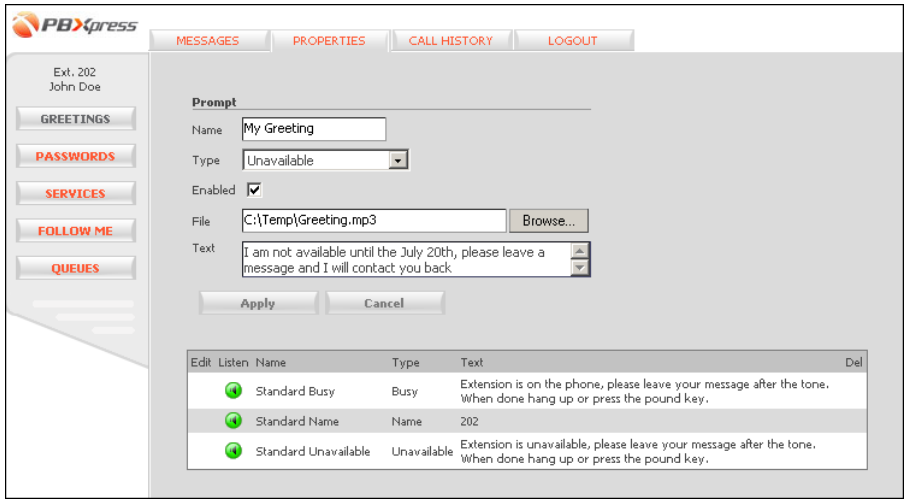
Properties

There are three types of parameters that users can customize:

Greetings

When a caller reaches your voicemail, they will hear an announcement inviting them to leave you a message. While some default prompts are supplied with PBXpress, users may, of course, upload their own prompts, recorded using their own voice.
To upload a new voice prompt, choose the **Properties** tab and then the **Greetings** button on the left.



Field	Description
Name	A name to distinguish this prompt from others.
Type	In which situation should this prompt be played? You can have different prompts played in the following situations: • Someone calls your extension, but your IP phone is offline or not answering – the **Unavailable** prompt is played. • Someone calls your extension, but you are currently on the phone – the **Busy** prompt is then played.
Enabled	Use this prompt for the specified type. You may upload multiple voice prompts, but only one of them will be used for a specific announcement type (e.g. Unavailable). The active (in use) voice prompt is marked by a green icon, while an unused prompt is marked by a yellow icon.
File	The path to the sound file. You may either type the

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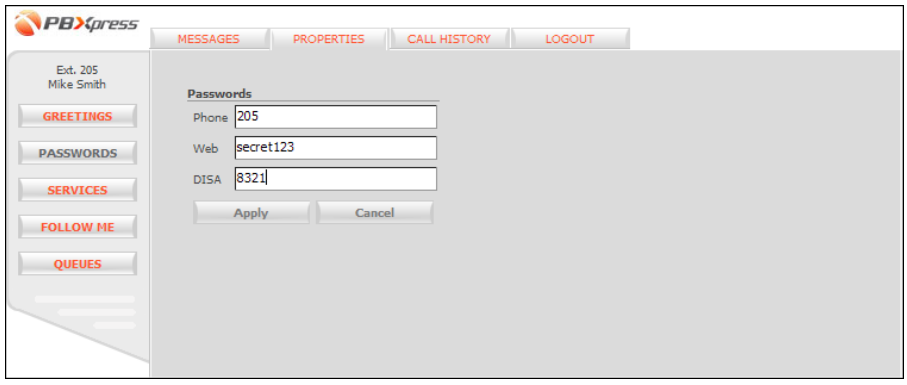
	path to the file manually, or use the **Browse** button to launch a file open dialogue.
Text	The text of the voice prompt. It is highly recommended that you enter the complete voice prompt text here, so that later you can find the required prompt by just a quick glance at the table of available prompts, instead of having to listen to every single one of them.

Passwords

For improved security, end users may change the passwords assigned to them by the system administrator. In fact, it is highly recommended that you change your password from the default one (identical to your extension number) immediately upon your first login.



Field	Description
Phone	The password you will use to access messages from an IP phone. Since this password must be typed on your phone's dial-pad, it must consist only of digits. If you enter an empty password, the system will not ask for a password when calling the voicemail access number from an IP phone.
Web	The password you will use to access messages from the web interface.
DISA	Password for accessing DISA service via the IVR.

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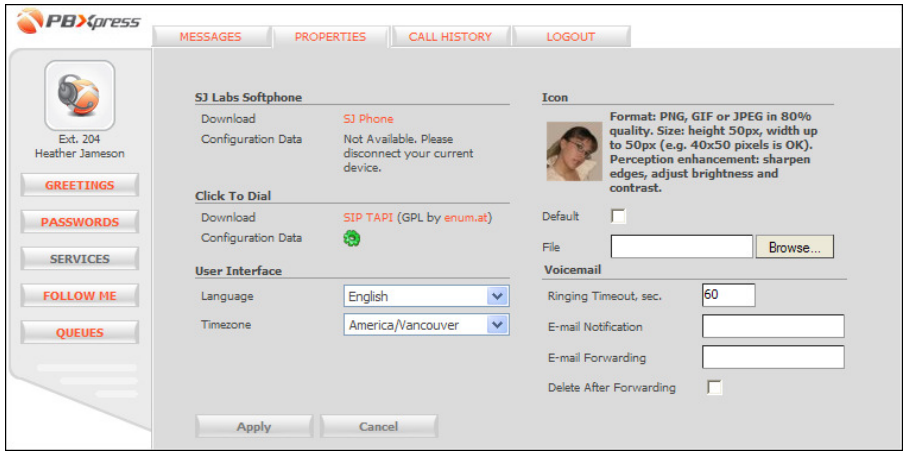
Services

You can change various service parameters, e.g. notification of new voice messages.



- **SJ Labs Softphone** section

Field	Description
Download	Click on the **SJ Phone** text to download and install the SJ Phone on your computer.
Configuration data	• - Downloads the SJ Phone configuration file for this extension. After your SJ Phone processes this file, it will automatically be configured for PBXpress. • - Downloads an SJ Phone configuration file which will remove this extension's data from the SJ Phone configuration.

- **Click To Dial** section

Field	Description
Download	Click on the **SIP TAPI** text to download and install the SIP TAPI client on your computer.
Configuration data	- Provides instructions on SIP TAPI configuration on your PC.

- **User Interface** section

Field	Description
Language	Specifies which language is used for this extension's self-care web pages.
Time Zone	Displays call times in the specified time zone.

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- **Icon** section

Field	Description
Icon	Upload a custom picture for easy identification of a user on the interface (especially the operator console).

- **Voicemail** section

Field	Description
Ringing Timeout, sec.	How long the phone will ring before the call is transferred to voicemail.
E-mail Notification	When a new voice message arrives, send a short notification to the specified email address.
E-mail Forwarding	Send a complete copy of every new voice message to the specified email address.
Delete After Forwarding	Do not leave a copy of the message on PBXpress when forwarding a message to an external email address. In this case, you will not be able to listen to this message from your IP phone or via the PBXpress web interface.

NOTE: PBXpress will only be able to deliver emails to an external email server if PBXpress administrator has properly configured the **SMTP Relay Host** parameter has been set up in the System/Services Configuration section.

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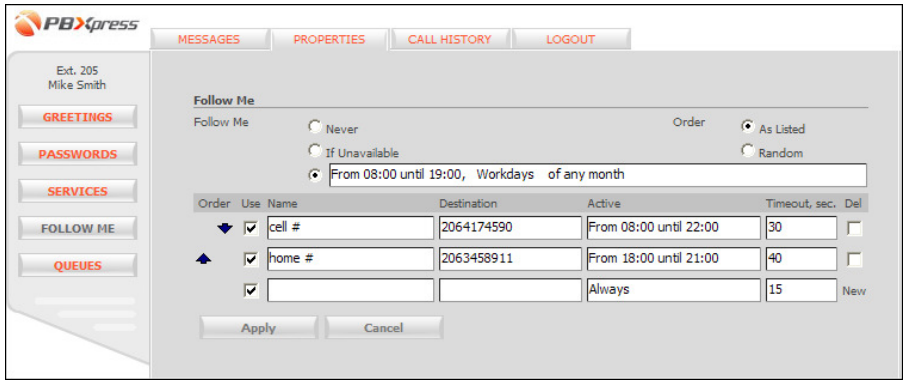
Follow-me

When it is impossible to deliver a call directly to an IP phone, it may be forwarded to another extension, or to a PSTN phone (mobile, office, home, etc.).



follow-me parameters:

Field	Description
Follow Me	• **Never** – Turn off the follow-me functionality. Useful if you want to temporarily disable follow-me for this particular account but keep all the other settings, such as the list of forward phone numbers. • **If Unavailable** – First try to ring on the IP phone. If not answered within a certain number of seconds, go to the follow-me numbers. • The last option allows you to specify a time period. Within the specified interval, PBXpress will not even try to ring on the SIP phone, but will immediately forward the call to one of the follow-me numbers. Useful when you know that you will not be able to answer your IP phone at a certain time (e.g. at night, or when leaving on vacation).
Order	• **As Listed** – Ring on the follow-me numbers sequentially, in the order they are listed on the web interface. • **Random** – Shuffle the list of numbers each time to produce a new sequence.

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Follow-me number configuration:

Field	Description
Order	Use the arrow buttons to change the order of the follow-me numbers in the list.
Use	Allows to quickly disable forwarding to this particular destination, but keeps all the other information (phone number, period) intact. Useful when you do not want to use this phone right now (e.g. your overseas cell phone), but you may need it again in the future (during your next business trip).
Name	Easy-to-understand identification of the destination (e.g. My Mobile).
Destination	Phone number the call should be forwarded to. The number must be entered in exactly the same format as it would be dialed from an IP phone. Thus, if you want to forward the call to extension 121, enter "121"; if you want to forward it to your overseas mobile, which you would dial as 011420732813200, then that is the number you should enter.
Active	By default, forwarding will work 24 hours a day (**Always**). If you wish to change this, click on the value in this column. The period wizard will appear, allowing you to precisely define when calls should go to this number.
Timeout, sec.	How long PBXpress should wait for someone to answer this phone number before moving on to the next number in the follow-me list.

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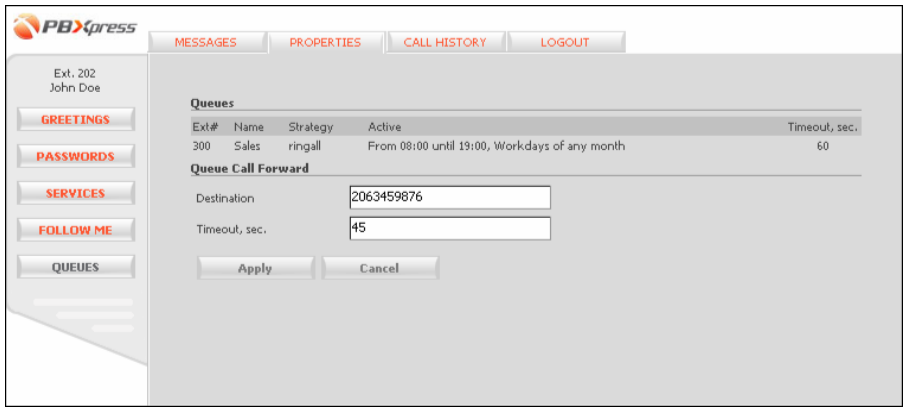
Queues

In order to provide effective distribution of calls in the queue, the follow-me service is not engaged when a call is forwarded to an extension via a queue. However, if you are not available on your IP phone, but still want to receive calls coming to you via a queue, you can specify a phone number to which these calls should be forwarded. For your convenience, this screen shows you a list of all the queues which your extension is part of.



NOTE: When you set up this parameter, your IP phone will not ring for incoming calls from a queue; rather, these calls will automatically go to the specified phone number.

Queue parameters:

Field	Description
Destination	The phone number calls should be forwarded to.
Timeout, sec.	How long PBXpress will wait for you to answer a call on this number.

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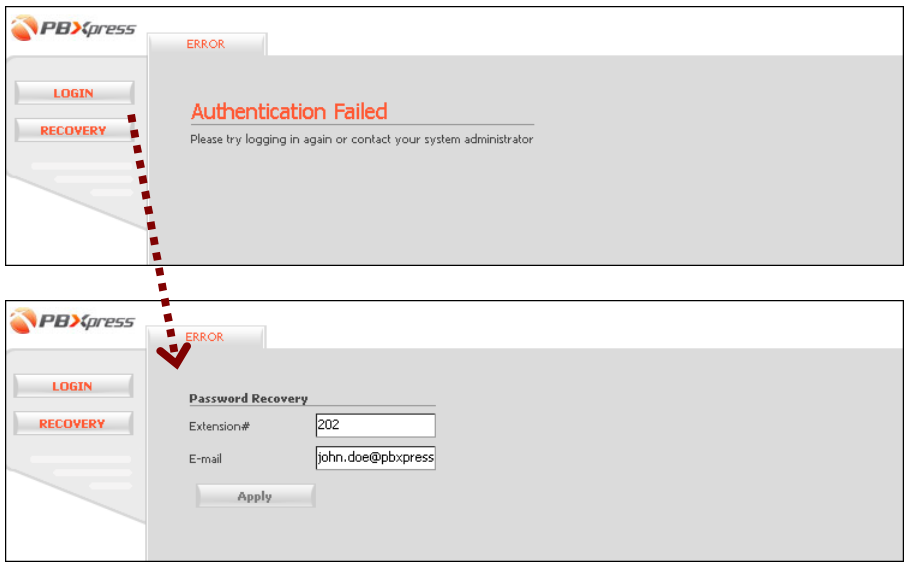
Recovering a Lost Extension Self-care Password

If, for some reason, the password for an extension's web access is no longer available, you can still restore it. After an unsuccessful login to the self-care web interface, you will see an Authentication Failed screen similar to the one below. After this, click on the **Recovery** button and enter an extension number and email address. PBXpress will generate a new password and send it to the specified email address.



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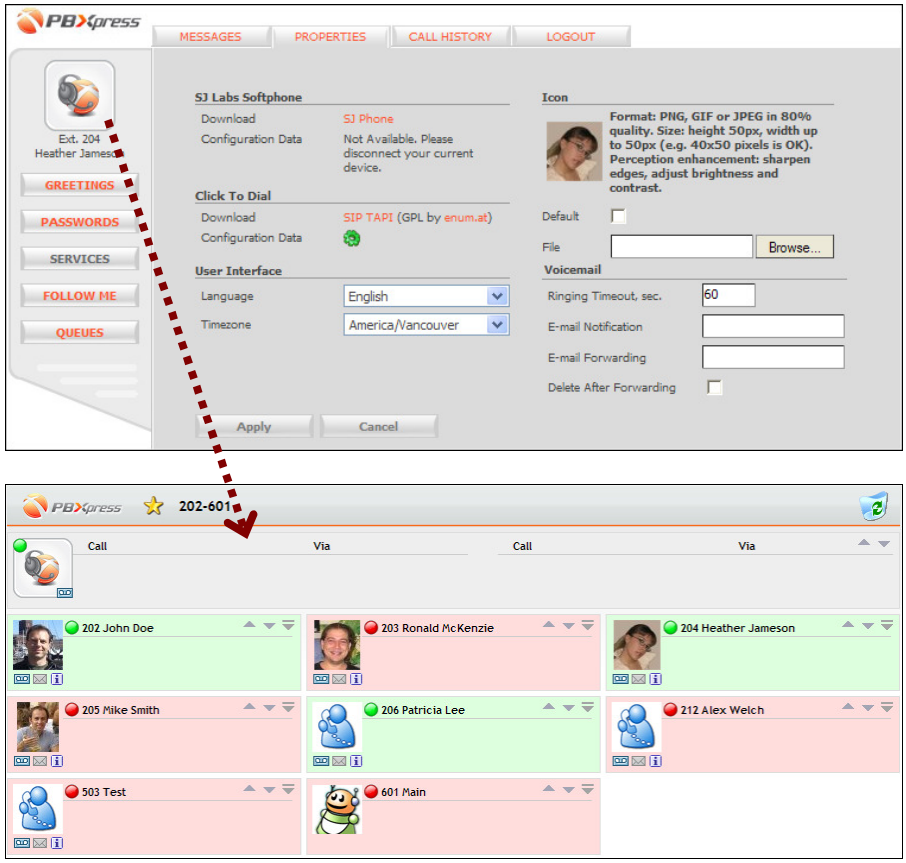
Operator Console

If your extension class allows access to the operator console, the corresponding icon will be shown in the upper-left corner. Click on it to launch the operator console.

Currently, the operator console requires Internet Explorer 7.0, and is not available for other browsers.



The operator console allows you to monitor all activities by users in real time (which extensions are online, who is on the phone right now, which calls are connected, etc.) and perform operations such as call transfer, hold or disconnect. The operator console will update the information **automatically**; please do not press "Reload" in your browser.

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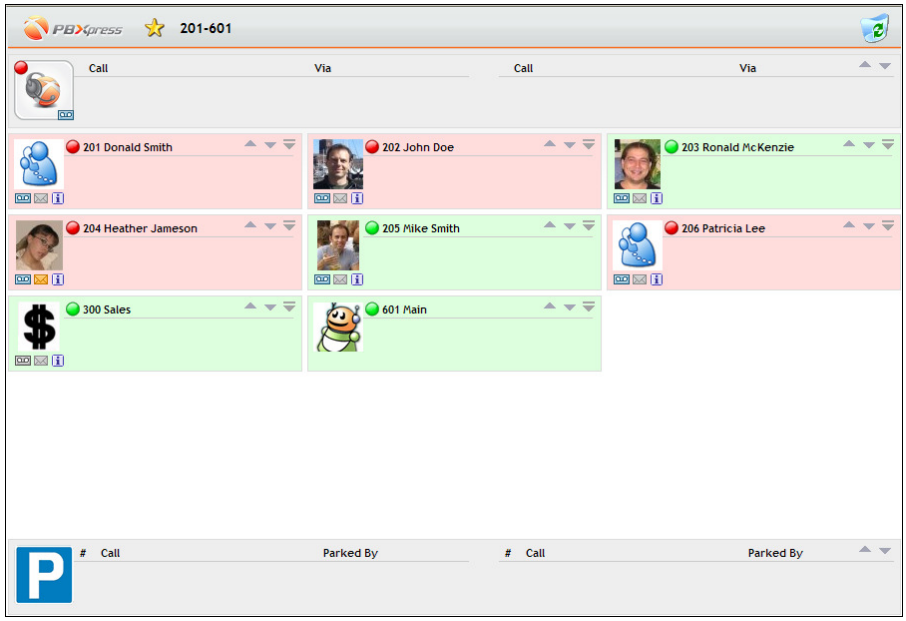
Operator console structure

The operator console runs in a separate web browser window, and contains several areas:



- On top are the Favorites ( ) icon, the extension index (202-601 in this example) and the Trash ( ) icon.
- Directly below this is information about the extension using the console and all calls associated with this extension.
- Finally, you will see all the entities in the current PBX admin group.
- At the very bottom you will see the parking area, which is used to manage parked calls.

Extension ranges and favorites

If there are several dozen extensions, queues and auto-attendants, it may be difficult to navigate between all of them. If all the elements do not fit onto a single screen, the extension index will contain multiple entries, e.g. **202 – 220, 221 – 240**, etc. Click on an index entry to display all the extensions in the given range.

Fortunately, in most cases only a small portion of the extensions require constant attention by the operator. These may be put in the Favorites list for easy access. To add an extension to Favorites, simply drag and drop its icon picture onto the icon. After this, when you click on the icon only your "favorite" extensions will be displayed.

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To remove an extension from the Favorites list, drag and drop it onto the Trash ( ) icon.

Operator info

The icon on the left displays the status of the operator's extension. If the circle in the upper-left corner is green, the extension is currently available. If the icon is red, the extension is currently busy.

The tape icon in the bottom-right corner represents a voicemail. In order to transfer a call to the operator's voicemail, simply drag and drop the call onto the tape icon.

A list of current calls will be displayed to the right of the operator's extension icon. The phone number is displayed for each call, while the connection name is given in the **Via** column for calls coming or going from outside PBXpress.

Call progress icons

There are several icons that indicate the current call status. The direction arrow tells whether a call is incoming or outgoing for this extension. An arrow facing the extension icon represents an incoming call, while an arrow facing away from it represents an outgoing call.

Icon	Description
	Connected call.
	(Blinking icon) Call on hold.
	Call in progress (not connected yet).
	(Blinking icon) A call is alerting (ringing) on this extension.
	Unknown call type.

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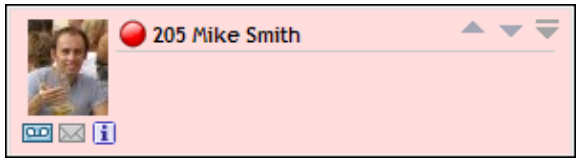
Extension info

Information about each extension is represented in an info area similar to the one below. Extensions which are currently offline are displayed on a pink background:



In the case of queues, a pink background means that none of the extensions allocated to this queue are currently online. For an attendant, a pink background means that the attendant is not currently active as per its schedule definition.

A circle next to the person's name indicates whether this extension is currently on the phone (red) or available (green).

The tape icon in the bottom right corner represents the voicemail box for this extension. A blue icon (as displayed on the picture above) means that this extension has the voicemail service activated. A gray icon ( ) indicates that voicemail is disabled for this extension.

The envelope icon indicates whether this user has an assigned email address in PBX ( ), or whether there is no known email address for this user ( ). If an email address is available, click on this icon to automatically start a new message to this person in your email client.

The information icon ( ) is reserved for future use.

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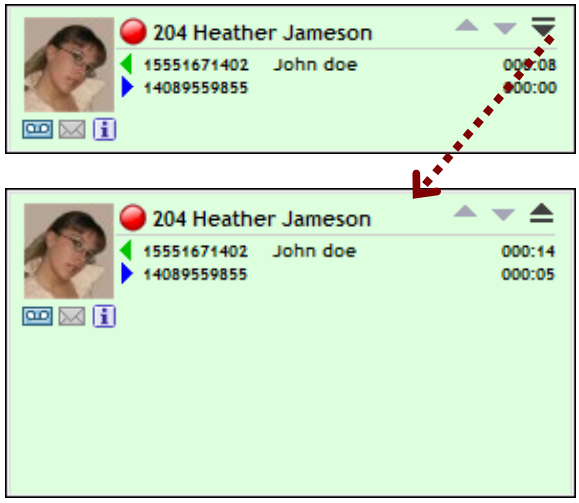
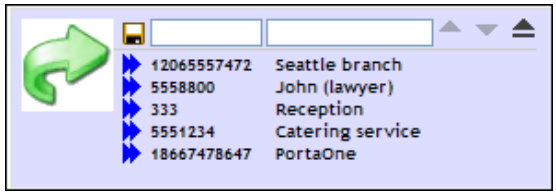
Current calls of an extension and their duration are displayed below the extension's name. Five calls are displayed by default; if the extension has more calls associated with it, you can scroll through the call list using the up/down arrows in the upper-left corner. Or you can expand this extension's profile into "wide" mode (available when browsing "Favorites" only).



Dial-pad

One element is always present on the Favorites screen – the dial-pad. Here you can enter a set of predefined phone numbers and their descriptions, so as to quickly forward calls to destinations outside your IP PBX.



To add a new number, simply type in the phone number and description and click the icon to save. To edit an existing entry, click the icon next to the phone number. To delete an entry, drag and drop it onto the Trash icon.

Now if, for example, you want to transfer a call to your travel agent, you can simply drag and drop it to the appropriate row in the dial-pad number list.

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Call control

- To disconnect a call, drag and drop the call onto the Trash icon.
- To transfer a call to another extension, drag and drop it onto the extension's icon. (You can transfer calls to queues or attendants in the same way.)
- To transfer a call to an extension's voicemail, drag and drop it onto the tape icon below the extension's icon.

Parking Area

In the parking area you will be able to see all calls, which currently have been parked by people in the same administrator group.

For each call the following information is displayed:

- Retrieval code (which should be dialed from another phone to retrieve the call)
- Identification of the party on hold (phone number and name when applicable)
- Current call duration
- Identification of the person, who put the call on hold

In order to park a call, drag-and-drop the call representation to the "parking" icon. Parked call can also be connected to a specific extension by performing drag-and-drop to that extension icon.

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Checking Voice Messages from an IP Phone

Dial the voicemail access number. If prompted, enter the voice mailbox password. The following voice menu is available:

```
- 1 Read voicemail messages
 - 3 Advanced options
   - 1 Reply
   - 3 Envelope
   - 5 Leave a message in another mailbox
   - * Return to the main menu
 - 4 Play previous message
 - 5 Repeat current message
 - 6 Play next message
 - 7 Delete/Undelete current message
 - 8 Forward message to another mailbox
 - 9 Save message to folder
 - * Help; during message playback: Rewind
 - # Exit; during message playback: Skip forward

- 2 Change folders
 - 0 New
 - 1 Old
 - 2 Work
 - 3 Family
 - 4 Friends
 - # Cancel

- 3 Advanced options
 - 5 Leave a message
 - * Return to the main menu

- 0 Mailbox options
 - 1 Record your unavailable message
 - 2 Record your busy message
 - 3 Record your name
 - 4 Change your password
 - * Return to the main menu

- * Help
- # Exit

While listening to a recorded voicemail message: Press # to
fast forward, or * to rewind by 3 sec. increments.
```

Using PBX Features from an IP Phone

Special features such as call transfer require support on the SIP device side. While IP phones normally have dedicated buttons for such functions, on analog telephone adapters you will have to use special

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combinations of normal dial keys (e.g. #90). Moreover, specific usage details (such as which keys should be pressed and in which sequence) are vendor-dependent, and may differ from one phone to another. Obviously, this is very inconvenient for the end user, so PBXpress has tried to improve the situation by provisioning analog telephone adapters to use the same keys, whenever possible. Of course, this only applies to supported SIP devices that are auto-provisioned by PBXpress.

Putting the Other Party on Hold

Sipura Adapters & Cisco ATA

Step 1 Press the flash button on the telephone handset to put the other party on hold.
Step 2 If necessary, you may now dial another extension and have a conversation (consultation hold).
Step 3 When you wish to resume the first conversation, press the flash button again.

Fully Unattended Transfer (Blind Transfer)

Sipura Adapters & Cisco ATA

Step 1 Press the flash button on the telephone handset to put the other party on hold and get a dial tone.
Step 2 Type #90 (transfer service activation code) on your telephone keypad, then enter the phone number to which you want to transfer the other party, then press #.
Step 3 Hang up your phone.

Attended Transfer (Consultative Transfer)

Cisco ATA

Step 1 Press the flash button on the telephone handset to put the other party on hold and get a dial tone.
Step 2 Dial the telephone number to which you want to transfer the other party.
Step 3 Talk to the other party first.
Step 4 Hang up your phone to transfer the other party.

Sipura Adapters

Step 1 Press the flash button on the telephone handset to put the other party on hold and get a dial tone.

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Step 2 Enter the phone number which you want to transfer the other party to, then press #.
Step 3 Wait until the call is connected, then talk to the other party before proceeding to the next step.
Step 4 Press the flash button to bridge the calls.
Step 5 Hang up your phone.

3-way Conference

Sipura Adapters & Cisco ATA

Step 1 Press the flash button on the telephone handset to put the other party on hold and get a dial tone.
Step 2 Dial the telephone number which you would like to join the conference.
Step 3 Wait for an answer, then press the flash button again in order to conference the call.

Retrieval of a parked call

Sipura Adapters & Cisco ATA

Step 1 Park the call using the attendant console, note the retrieval code in the leftmost column.
Step 2 Dial this code from any IP phone in a same way you would dial an extension number.

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Performing a Firmware Update

Main Concepts

A dedicated team of PBXpress developers is constantly working on improving the software and adding new features to it. If you installed your PBXpress system some time ago, you may well be interested in the latest improvements. The PBXpress team has developed a system of automated updates in order to allow you to make use of the latest features without interrupting service (e.g. sending your PBXpress box to our office for an upgrade).

The update process may be divided into several steps:

1. PBXpress releases a new firmware version and publishes it on the PBXpress website. (Firmware versions are distinguished by their publication date and a special version number)
2. Your PBXpress checks whether a new firmware version is available.
3. A new firmware image is downloaded (either automatically or by an admin command) to PBXpress and becomes available for activation.
4. You can now choose to activate the new firmware; this will involve a PBXpress reboot.
5. After this, the new firmware is ready for use (the previous version is not erased, and you can always roll back to it).

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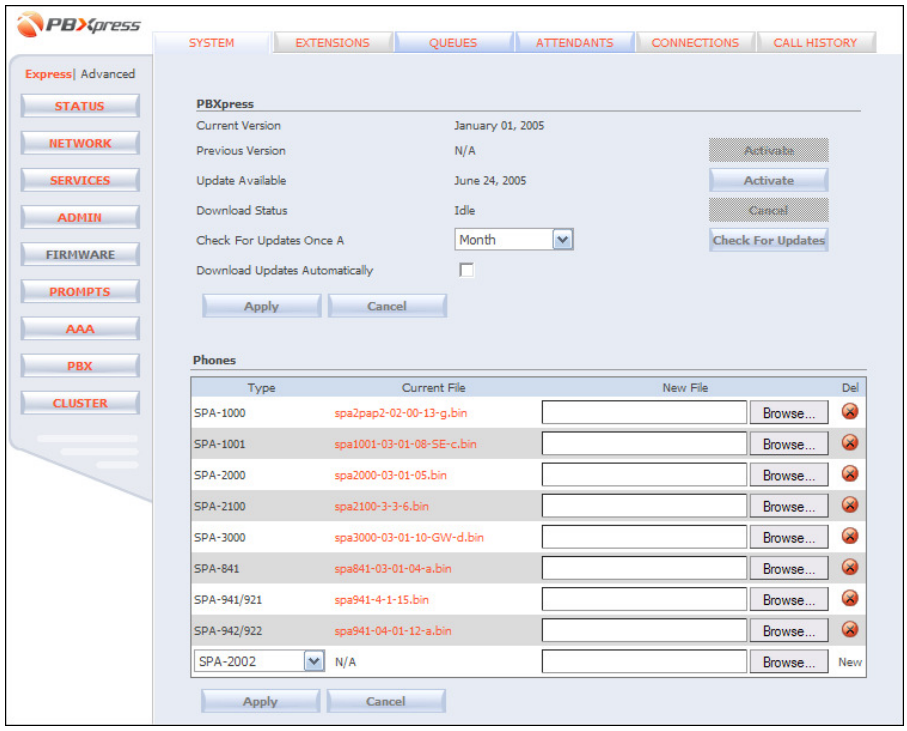
Configuration



Field	Description
Current Version	Firmware version currently active.
Previous Version	After a firmware update, the ID of the stable version which was used before the update is retained, so you can roll back to it in case of any problems with the new firmware.
Update Available	The most recent firmware downloaded locally from the PBXpress site and available for activation.
Download Status	Displays progress info during the firmware download process.
Check For Updates Once A	Select the desired period for contacting the PBXpress website.
Download Updates Automatically	If this option is on, PBXpress will download new firmware immediately after it becomes available. If off, the administrator must manually initiate a firmware download.

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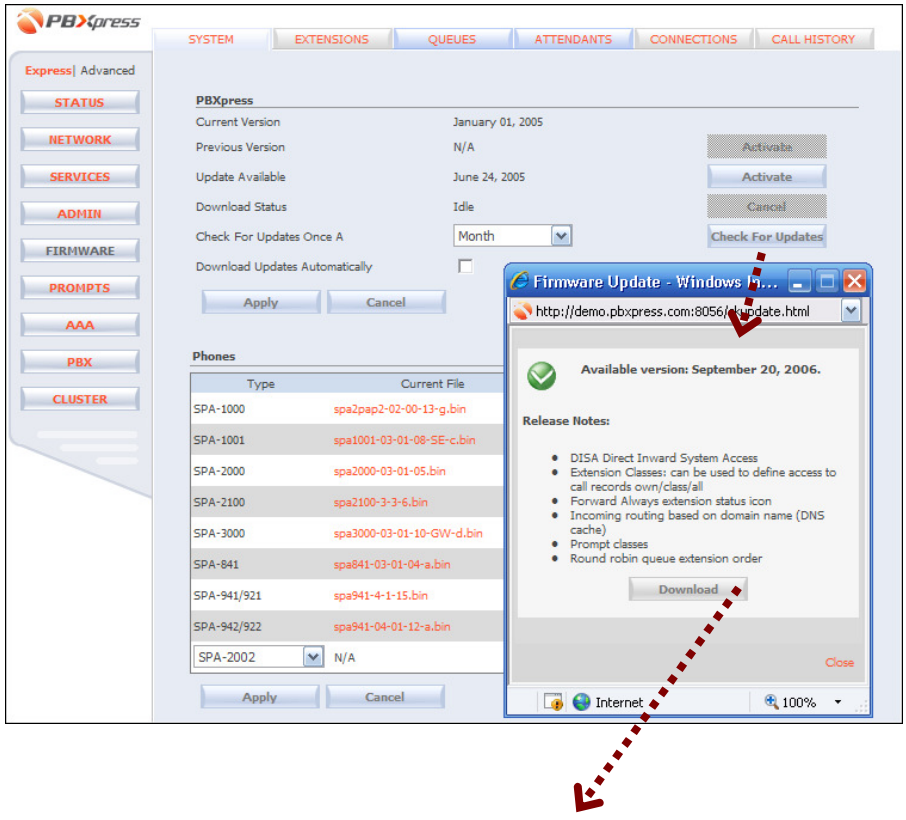
Firmware Download

PBXpress will periodically check the PBXpress website to see if a new update is available. If you have not activated the Automatic Update Download option, PBXpress will not download the image, but instead send a notification to the administrator (to the email address defined in the **System/Admin** section), so that he can initiate the download manually. This may be useful if the administrator prefers to handle such downloads in off-peak hours only, when they will not affect Internet bandwidth usage. If you have decided to use automated update downloads, PBXpress will download the update to local storage and then send a notification so that the administrator can initiate the upgrade. The administrator can always check for available updates manually, as shown on the next page.



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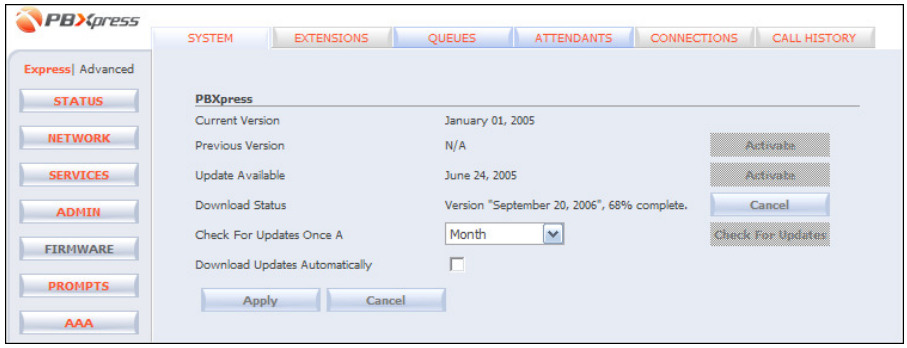

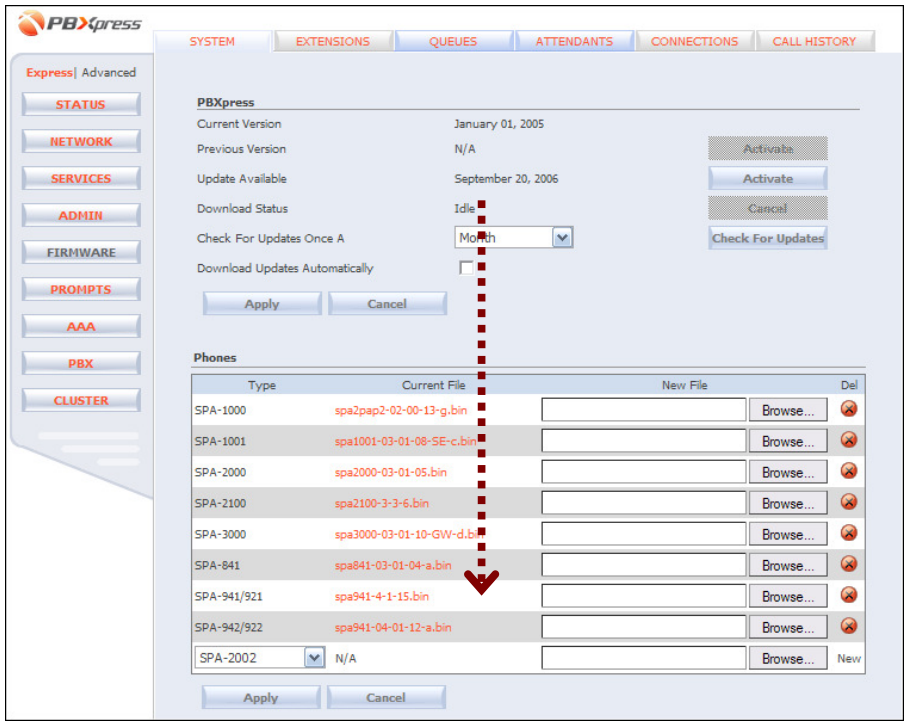

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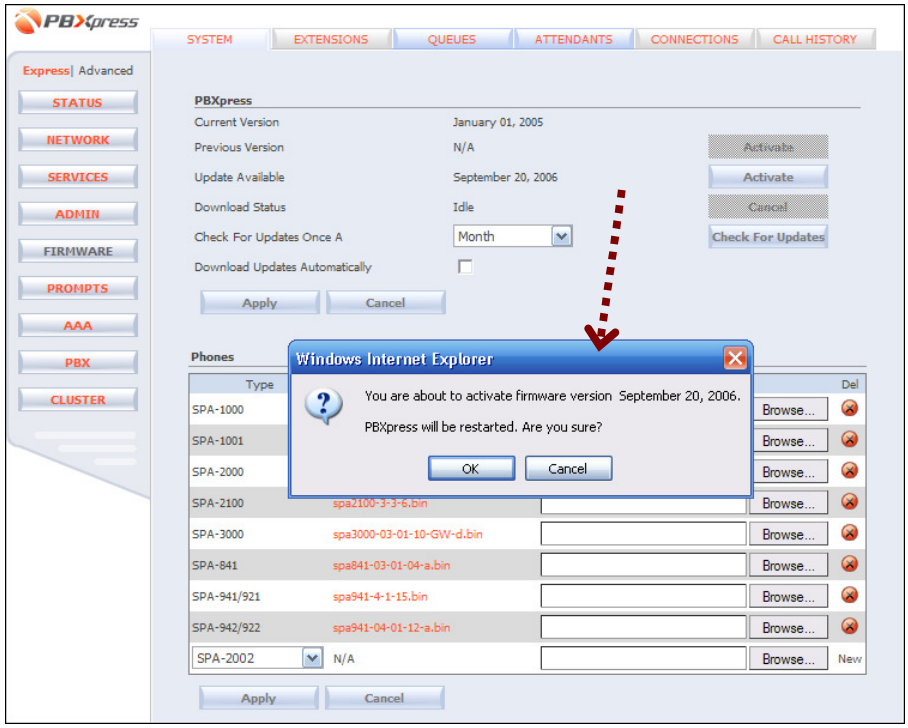
Switching to a New Firmware Version

In order to activate the newly downloaded firmware, simply press the **Activate** button next to the firmware version.



If you press **OK**, the new firmware version will be installed, and PBXpress will reboot. Wait 2 to 3 minutes, then try to log in to the PBXpress administrator interface again.

If PBXpress is unable to boot after a firmware update, just press the **Reset** button on the PBXpress box. PBXpress will automatically roll back to the previous firmware version.

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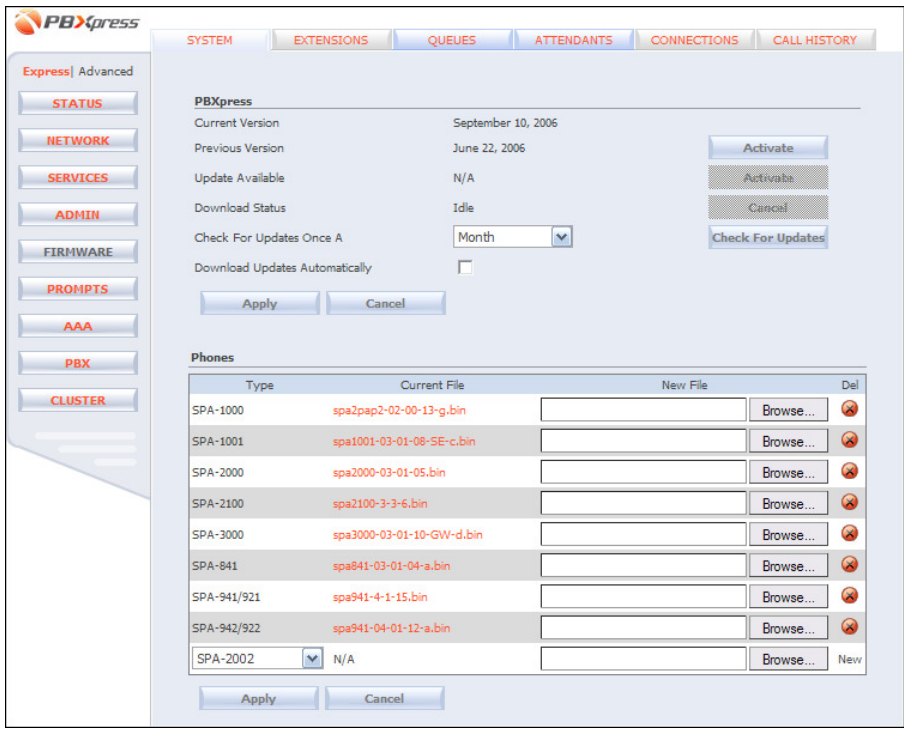
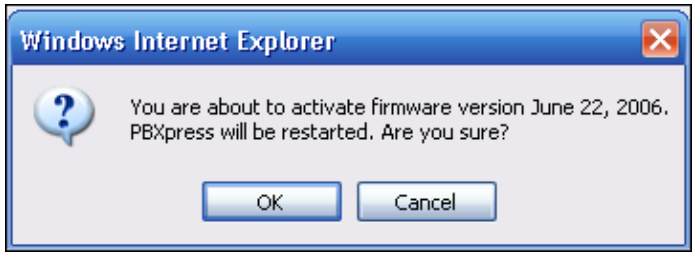
After Upgrading

After a successful upgrade, you can check the upgrade status in the **System / Firmware** administration panel.



The **Current Version** field shows the actual firmware version. Note that an **Activate** button is now available next to **Previous Version**. If you click on it, you will receive a prompt like the one below and, upon confirming that you wish to restore the old firmware, PBXpress will reboot and the previous firmware version will be restored.



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Frequently Asked Questions

Are there any laws which regulate call recording?

Generally, it is legal to record any conversation where all the parties to it have given their consent (one-party consent if all the parties are in a state with a corresponding law). U.S. federal law requires only one-party consent to the recording of a telephone conversation, but does not explicitly protect such a recording if it is made for criminal or tortious purposes. Many states make similar exceptions.
You can obtain more information on this subject from:

- EU Telecommunications Data Protection Directive
- Infowar.com's Law Section: Computer Privacy Community **www.infowar.com/iwftp/cpd/CPD-Library/**, **www.infowar.com**
- State-by-state summaries: **www.rcfp.org/taping/states.html**

If I only connect PBXpress to LAN, can I still use VoIP services?

Yes, assuming that:

1. You are using a symmetric NAT on your network which does not block outgoing UDP connections from the internal network.
2. The remote SIP server supports NAT traversal for signaling and media traffic.

In that case, PBXpress will be able to communicate with remote SIP servers just fine, so you will be able to make outgoing and receive incoming calls.

How can an employee use his IP phone while connected to the Internet outside the office network (LAN)?

In order to provide this sort of service, PBXpress must utilize the WAN interface, so a public IP address will be available. The only change required in the IP phone configuration is that the IP phone must use this IP address as the address of the SIP server. Currently, this configuration

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has to be done manually for all IP phones except Cisco ATA , so the best way is as follows:

1. Connect the IP phone to LAN.
2. Provision it using the PBXpress configuration tool.
3. Consult the information in the pop-up window which appears during provisioning. This will give you more information about the IP phone and how it can be configured manually. Alternatively, consult the manual available from the IP phone's vendor.
4. Manually change the address of the SIP server in the IP phone's configuration.
5. The phone can now be plugged into an external network (e.g. behind the user's home DSL router).

How can a user seamlessly move his phone between the office network (LAN) and his home network?

If a user wishes, he can take his work phone home in the evening, and plug it in there and use it (without any configuration changes), and then perform the reverse operation in the morning. This is possible if the IP phone supports the entry of two SIP servers (a main one and a backup one). The only required change in the IP phone configuration is keeping your PBXpress LAN IP address as the main SIP server address, and adding the PBXpress WAN IP address as a backup SIP server. See the previous question for instructions on how to manually alter an IP phone's configuration.

How can I use PBXpress in a multi-office environment?

If you have multiple offices, each one of which is equipped with PBXpress, it is very easy to integrate them into a single "virtual" office environment. The pre-requisites are as follows:

- Make sure you follow a numbering scheme for extensions throughout your organization; for example, all extensions from office A will start with 2 (e.g. 201, 202, etc.), extensions from office B start with 3 (301, 302, etc.), and so forth.
- If you are not using a central SIP server (e.g. PortaSwitch), then every PBXpress must be connected to WAN and available at a certain public IP address.

Then the only required configuration step on PBXpress in office A is to create a VoIP connection routing all calls to 3.. numbers to the IP address of PBXpress in office B, and vice versa for office B to office A. Note that the E.164 pattern in this case must be 3.. (and not 3*).

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How many simultaneous calls can I have on my LAN (PBXpress)?

Please see the table below regarding the required capabilities.

Cable Capacity	Bandwidth %	Phone Usage at G.711	"Voice Channels" G.711	"Voice Channels" G.729 (x 2.5)
10BaseT Half	40%.	2%	20	50
10BaseT Full	80%	1%	80	200
100BaseT Half	40%	0.2%	200	500
100BaseT Full	80%	0.1%	800	2000

How much Internet bandwidth do I need for my VoIP calls?

Obviously this will depend on the codec being used. When calculating the actual amount of IP bandwidth, you must also remember that every RTP packet has an IP header in addition to the actual data (payload); this header must be transferred along with the data, and therefore requires a certain amount of bandwidth. Since RTP (voice stream) makes use of many small packets, the amount of bandwidth used by IP headers becomes unexpectedly significant. See the table below for estimated bandwidth figures.

Codec	Bit rate	Nominal bandwidth
G711	64 Kbps	87.2 Kbps
G729	8 Kbps	31.2 Kbps
G723.1	6.4 Kbps	21.9 Kbps
G723.1	5.3 Kbps	20.8 Kbps

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Appendixes

Appendix A – Glossary

ANI - Automatic Number Identification. A feature in which a series of digits, either analog or digital, are included in a call, thus identifying the caller's telephone number.

ATA - Analog Telephone Adapter.

BRI - Basic Rate Interface. BRI consists of two 64 kb/s B channels and one 16 kb/s D channel for a total of 144 kb/s. This basic service is intended to meet the needs of most individual users. BRI is used predominantly in Europe.

CLI - Calling Line Identification. A range of services in which a caller's number can be accessed by the recipient.

DHCP - Dynamic Host Configuration Protocol. A protocol for assigning dynamic IP addresses to devices on a network. In some systems, the device's IP address can change even while it is still connected.

Dialing rules - A set of definitions as to how a phone number should be presented (for example, should it include a country code or not).

DID - Direct Inward Dialing. Connecting an incoming call to the destination based on the dialed number.

DISA - Direct Inward System Access. A feature on PBX's which allows an external caller access to the PBX and all of its features such as tie and watts lines by dialing in on the DISA trunk.

DNS - Domain Name System (or Service). An Internet service that translates domain names to or from the IP addresses which are the actual basis of Internet addresses.

DSL - Digital Subscriber Line. A technology for bringing higher-bandwidth information to homes and businesses over ordinary copper telephone lines.

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E1 – A wide-area digital transmission scheme, used predominantly in Europe, that carries data at a rate of 2.048 Mbps. E1 lines can be leased for private use from common carriers.

Extension - An additional telephone set that is connected to the same telephone line.

Firmware - Software routines that are permanently written onto read-only memory.

FXO - Foreign Exchange Office. An interface that connects to the Public Switched Telephone Network's (PSTN) central office, and is offered on a standard telephone. Example: the RJ-11 connector that enables an analog connection to the central office.

HDD - Hard Disk Drive.

IP telephony - IP (Internet Protocol) telephony is a technology that supports voice, data and video transmission via IP-based LANs, WANs, and the Internet. This includes VoIP (Voice over IP).

ISDN - Integrated Services Digital Network. A service offered by most telephone carriers for the transmission of voice and data.

ISP - Internet Service Provider. A company that provides access to the Internet through modems, ISDN, T1s, and the like.

LAN - Local Area Network. This may be a network connecting a company's computers.

PBX - Private Branch Exchange. A privately-owned telephone switch, often used in large corporations to provide inside telephone connectivity and access to the PSTN.

PCM - pulse code modulation. Audio sent as a digital pulse code modulation (PCM) stream eliminates problems due to audio levels and attenuation.

PSTN - Public Switched Telephone Network. Also know as Plain Old Telephone System, this refers to the world's interconnected public telephone networks designed primarily for voice traffic.

RADIUS - A client/server protocol and software that enables remote access servers to communicate with a central server in order to authenticate dial-in users and authorize their access to a requested system or service.

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RFC - Request For Comments. An RFC is a standard document describing protocols, systems, or procedures used by the Internet community.

SIP - Session Initiation Protocol. SIP is an Internet Engineering Task Force (IETF) standard protocol for initiating interactive user sessions involving multimedia elements such as video, voice, chat, gaming, and virtual reality.

SMTP - Simple Mail Transfer Protocol. A protocol used to send email on the Internet. SMTP is a set of rules regarding the interaction between a program sending email and a program receiving email.

T1 - A type of data connection able to transmit a digital signal at 1.544 megabits per second. T1 lines are often used to link large computer networks, such as those that make up the Internet.

VoIP - Voice over Internet Protocol. The use of packet-based networks to send voice data, instead of the standard public switched telephone network.

WAN - Wide Area Network. A network connecting computers within very large areas, such as states, countries, or the world.

WAV - The WAVeform sound format. Microsoft's format for encoding sound files, which therefore carry the .wav extension.

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Appendix B – Supported IP Phones, Softphones, Analog Phone Adapters and Gateways

A PBXpress compatible phone, adaptor or gateway must satisfy the following requirements:

1. It must use SIP protocol.
2. It should have a G729 codec.

Please visit **http://www.pbxpress.com/user_agents.htm** for obtaining up to date list of compatible hardware and software.

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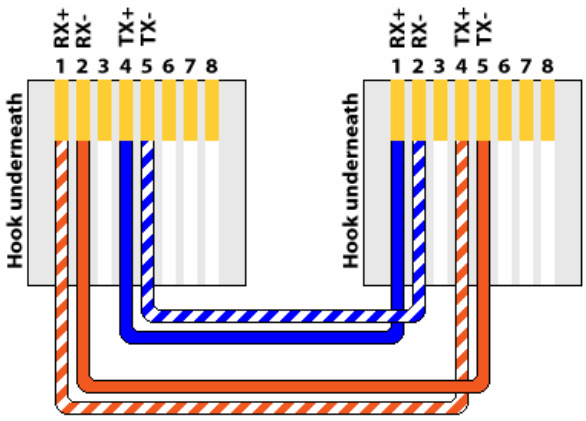
Appendix C – How to Make a Crossover T1/E1 Cable

If you need to connect PBXpress with some other device (e.g. a legacy PBX or Cisco router) via an E1 or T1 interface, you will need a special "crossover" cable. This is because the normal cable is designed to connect an end-point device (such as PBX or a gateway) and the network side. In our case, however, we have two end-points, so the cable has to be adjusted accordingly to allow them communicate correctly. See the diagram below for the correct pin-out.



PRI (T1/E1) CrossOver/Loopback Cable

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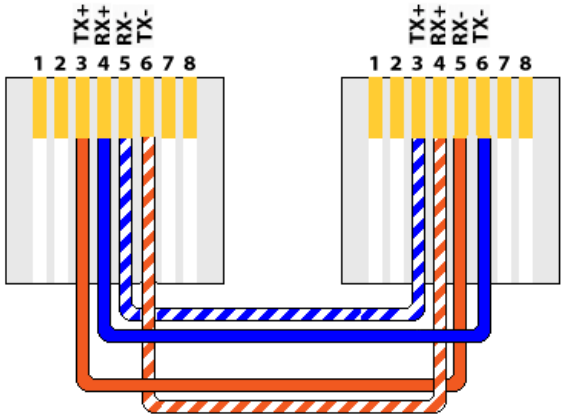
Appendix D – How to Make a Crossover BRI Cable

When the BRI port is run in the network mode, and connected to an ISDN terminal device, a BRI crossover cable is needed to connect the transmit and receive signal pairs between the two devices. The pinouts for a suitable RJ45 ISDN BRI crossover cable are shown here:



BRI CrossOver Cable

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